Logic Express 9

Instruments

Copyright © 2009 Apple Inc. All rights reserved.

Your rights to the software are governed by the accompanying software license agreement. The owner or authorized user of a valid copy of Logic Express software may reproduce this publication for the purpose of learning to use such software. No part of this publication may be reproduced or transmitted for commercial purposes, such as selling copies of this publication or for providing paid for support services.

The Apple logo is a trademark of Apple Inc., registered in the U.S. and other countries. Use of the "keyboard" Apple logo (Shift-Option-K) for commercial purposes without the prior written consent of Apple may constitute trademark infringement and unfair competition in violation of federal and state laws.

Every effort has been made to ensure that the information in this manual is accurate. Apple is not responsible for printing or clerical errors.

Note: Because Apple frequently releases new versions and updates to its system software, applications, and Internet sites, images shown in this manual may be slightly different from what you see on your screen.

Apple
1 Infinite Loop
Cupertino, CA 95014
408-996-1010
www.apple.com

Apple, the Apple logo, GarageBand, Logic, Mac, and Ultrabeat are trademarks of Apple Inc., registered in the U.S. and other countries.

Finder is a trademark of Apple Inc.

Other company and product names mentioned herein are trademarks of their respective companies. Mention of third-party products is for informational purposes only and constitutes neither an endorsement nor a recommendation. Apple assumes no responsibility with regard to the performance or use of these products.

Contents

Preface 9 **An Introduction to the Logic Express Instruments**
9 About the Logic Express Instruments
10 About the Logic Express Documentation
10 Additional Resources

Chapter 1 13 **ES E**
14 Getting to Know the ES E Interface
15 Using the ES E Oscillators
15 Using the ES E LFO
16 Using the ES E Filter
17 Using the ES E Envelope
17 Using the ES E Output Parameters
18 Extended ES E Parameters

Chapter 2 19 **ES M**
20 Getting to Know the ES M Interface
21 Using the ES M Oscillator
22 Using the ES M Filter and Filter Envelope
23 Using the ES M Level Envelope and Output Controls
23 Extended ES M Parameters

Chapter 3 25 **ES P**
26 Getting to Know the ES P Interface
27 Using the ES P Oscillators
27 Using the ES P LFO
28 Using the ES P Filter
29 Using the ES P Envelope and Level Controls
30 Using the Integrated ES P Effects
30 Extended ES P Parameters

Chapter 4 33 **ES1**
34 Getting to Know the ES1 Interface
35 Using the ES1 Oscillators
37 Using the ES1 Filter Parameters

38 Using the ES1 Amplifier Parameters
39 Using the ES1 Envelope Parameters
41 Modulating the Sound of the ES1
44 Adjusting Global ES1 Parameters
45 ES1 MIDI Controller List

Chapter 5 47 **ES2**
48 ES2 Architecture and Features
49 Getting to Know the ES2 interface
50 Using the ES2 Oscillators
62 Using the ES2's Global Parameters
64 Using the ES2 Filters
74 Using the ES2 Amplifier Parameters
76 Working with Modulation in the ES2
106 Using the ES2's Integrated Effect Processing Section
108 Creating Random ES2 Sound Variations
110 Using Macro Controls and Assigning Controllers in the ES2
112 ES2 Tutorial: Creating Sounds
123 ES2 Tutorial: Using Templates to Create Sounds

Chapter 6 131 **EFM1**
132 Getting to Know the EFM1 Interface
133 Working with the EFM1 Modulator and Carrier Parameters
136 Working with the EFM1 Modulation Parameters
137 Adjusting Global EFM1 Parameters
138 Setting EFM1 Output Parameters
139 Creating EFM1 Random Sound Variations
139 Assigning EFM1 MIDI Controllers

Chapter 7 141 **EVOC 20 PolySynth**
141 How the EVOC 20 Polysynth Works
144 Getting to Know the EVOC 20 PolySynth Interface
145 EVOC 20 PolySynth Sidechain Analysis Parameters
147 EVOC 20 PolySynth (U/V) Detection Parameters
148 EVOC 20 PolySynth Synthesis Section Parameters
153 EVOC 20 PolySynth Formant Filter Parameters
156 EVOC 20 PolySynth Modulation Parameters
157 EVOC 20 PolySynth Output Parameters
158 Getting the Best Results with the EVOC 20 PolySynth
160 A Brief Vocoder History
163 EVOC20 Block Diagram

Chapter 8 165 **External Instrument**
165 Getting to Know the External Instrument Interface
166 Using the External Instrument

Chapter 9 167 **EXS24 mkII**
168 EXS24 mkII Features
169 Getting to Know the EXS24 mkII Interface
170 About EXS24 Sampler Instruments
172 Getting to Know the EXS24 mkII Parameter Window
173 Using the EXS24 mkII Sampler Instruments Pop-Up Menu
178 Adjusting EXS24 mkII Global Parameters
182 Using the EXS24 mkII Pitch Parameters
184 Working with EXS24 mkII Filter Parameters
187 EXS24 mkII Output Parameters
188 Working with EXS24 mkII Modulation
202 An Overview of the EXS24 mkII Instrument Editor
205 Creating EXS24 mkII Instruments, Zones, and Groups
210 Editing EXS24 mkII Zones and Groups
213 Setting EXS24 mkII Zone Parameters
215 Using the EXS24 mkII Zone Loop Parameters
216 Setting EXS24 mkII Group Parameters
218 Using the EXS24 mkII Advanced Group Selection Parameters
219 Graphical Editing of EXS24 mkII Zones and Groups
221 Saving, Renaming, and Exporting EXS24 mkII Instruments
222 Editing Samples in the EXS24 mkII Sample Editor
223 Using an External Instrument Editor with the EXS24 mkII
224 Importing EXS24 mkII Sampler Instruments
233 Managing EXS24 Sampler Instruments
234 Setting EXS24 mkII Sampler Preferences
237 Configuring EXS24 mkII Virtual Memory
239 Advanced EXS24 mkII RAM Management
239 Using the VSL Performance Tool in EXS24 mkII

Chapter 10 241 **Klopfgeist**
242 Using the Klopfgeist Parameters

Chapter 11 243 **Ultrabeat**
244 Getting to Know the Structure of Ultrabeat
246 Loading and Saving Ultrabeat Settings
247 Getting to Know the Ultrabeat Interface
247 Getting to Know Ultrabeat's Assignment Section
251 Importing Sounds and EXS Instruments into Ultrabeat

254 Getting to Know the Ultrabeat Synthesizer Section
256 Getting to Know the Ultrabeat Oscillators
257 Getting to Know the Ultrabeat Oscillator Parameters
258 Using Oscillator 1 in Ultrabeat
262 Using Oscillator 2 in Ultrabeat
267 Using the Ultrabeat Ring Modulator
268 Using the Ultrabeat Noise Generator
269 Getting to Know the Ultrabeat Filter and Distortion Section
270 Using Ultrabeat's Multimode Filter
272 Using Ultrabeat's Distortion Circuit
273 Using the Ultrabeat Output Section
279 Working with Modulation in Ultrabeat
289 Working with the Ultrabeat Step Sequencer
299 Automating Parameter Values in Ultrabeat's Step Sequencer
303 Exporting Ultrabeat Patterns as MIDI Regions
304 Using MIDI to Control Ultrabeat's Sequencer
305 Ultrabeat Tutorial: Introduction
305 Ultrabeat Tutorial: Creating Kick Drums
310 Ultrabeat Tutorial: Creating Snare Drums
316 Ultrabeat Tutorial: Creating Toms and Tonal Percussion
316 Ultrabeat Tutorial: Creating Hi-Hats and Cymbals
317 Ultrabeat Tutorial: Creating Metallic Sounds
318 Ultrabeat Tutorial: Creating Extreme Sounds
318 Ultrabeat Tutorial: Programming in Building Blocks

Chapter 12 321 **GarageBand Instruments**
322 GarageBand Instrument Features
323 GarageBand Analog Basic
324 GarageBand Analog Mono
325 GarageBand Analog Pad
326 GarageBand Analog Swirl
328 GarageBand Analog Sync
329 GarageBand Bass
330 GarageBand Church Organ
331 GarageBand Digital Basic
332 GarageBand Digital Mono
333 GarageBand Digital Stepper
334 GarageBand Drum Kits
335 GarageBand Electric Clav(inet)
335 GarageBand Electric Piano
336 GarageBand Guitar
337 GarageBand Horns
338 GarageBand Hybrid Basic

340 GarageBand Hybrid Morph
342 GarageBand Piano
343 GarageBand Sound Effects
344 GarageBand Strings
345 GarageBand Tonewheel Organ
346 GarageBand Tuned Percussion
347 GarageBand Voice
348 GarageBand Woodwind

Appendix 349 **Synthesizer Basics**
349 Sound Basics
353 What Is a Synthesizer?
355 How Subtractive Synthesizers Work
369 Other Synthesis Methods
373 A Brief History of the Synthesizer

An Introduction to the Logic Express Instruments

Logic Express provides a number of software-based instruments that include: innovative synthesizers, a powerful sampler, and authentic recreations of vintage instruments. You can use these instruments for real time playback in Logic Express.

The Logic Express instruments cover almost every sound generation need you will encounter in your day to day work with Logic Express. All instruments feature intuitive interfaces that provide access to all functions and parameters. Pristine audio quality is assured when needed, or—at the other end of the spectrum—extremely harsh and dirty sounds can be created with many of the included instruments. All instruments are highly optimized for efficient CPU usage, maximizing the real time playback potential of your computer.

This preface covers the following:

- About the Logic Express Instruments (p. 9)
- About the Logic Express Documentation (p. 10)
- Additional Resources (p. 10)

About the Logic Express Instruments

Logic Express includes a comprehensive collection of instruments that can be used in your music projects. Instruments are grouped in the following categories.

Instrument category	Included instruments
Synthesizer	• EFM1 • ES E • ES M • ES P • ES1 • ES2 • Klopfgeist
Drum synthesizer	• Ultrabeat
Software sampler	• EXS24 mkII
Vocoder synthesizer	• EVOC 20 PolySynth

Instrument category	Included instruments
Utility	• External Instrument
GarageBand instruments	Analog Basic, Analog Mono, Analog Pad, Analog Swirl, Analog Sync, Bass, Digital Basic, Digital Mono, Digital Stepper, Drum Kits, Electric Clavinet, Electric Piano, Guitar, Horns, Hybrid Basic, Hybrid Morph, Piano, Sound Effects, Strings, Tonewheel Organ, Tuned Percussion, Voice, Woodwind (see GarageBand Instruments)

About the Logic Express Documentation

Logic Express comes with various documents that will help you get started as well as provide detailed information about the included applications.

- *Logic Express User Manual:* This onscreen manual provides comprehensive instructions for using Logic to set up a recording system, compose music, edit audio and MIDI files, and output audio for CD productions.
- *Exploring Logic Express:* This booklet provides a fast-paced introduction to the main features and tasks in Logic Express, encouraging hands-on exploration for new users.
- *Logic Express Control Surfaces Support:* This onscreen manual describes the configuration and use of control surfaces with Logic Express.
- *Logic Express Instruments:* This onscreen manual provides comprehensive instructions for using the powerful collection of instruments included with Logic Express.
- *Logic Express Effects:* This onscreen manual provides comprehensive instructions for using the powerful collection of effects included with Logic Express.
- *Logic Express Working with Apogee Hardware:* This onscreen manual describes the use of Apogee hardware with Logic Express.

Additional Resources

Along with the documentation that comes with Logic Express, there are a variety of other resources you can use to find out more.

Release Notes and New Features Documents

Each application offers detailed documentation that covers new or changed features and functions. This documentation can be accessed in the following location:

- Click the Release Notes and New Features links in the application Help menu.

Logic Express Website

For general information and updates, as well as the latest news on Logic Express, go to:

- http://www.apple.com/logicexpress

Apple Service and Support Websites

For software updates and answers to the most frequently asked questions for all Apple products, go to the general Apple Support webpage. You'll also have access to product specifications, reference documentation, and Apple and third-party product technical articles.

- http://www.apple.com/support

For software updates, documentation, discussion forums, and answers to the most frequently asked questions for Logic Express, go to:

- http://www.apple.com/support/logicexpress

For discussion forums for all Apple products from around the world, where you can search for an answer, post your question, or answer other users' questions, go to:

- http://discussions.apple.com

ES E

1

The 8-voice ES E (ES Ensemble) synthesizer is ideal for quickly creating warm, rich pad and ensemble sounds.

ES E produces sounds using subtractive synthesis. It features an oscillator that generates harmonically rich waveforms. You *subtract* (cut, or filter out) portions of these waveforms, and reshape them, to create new sounds.

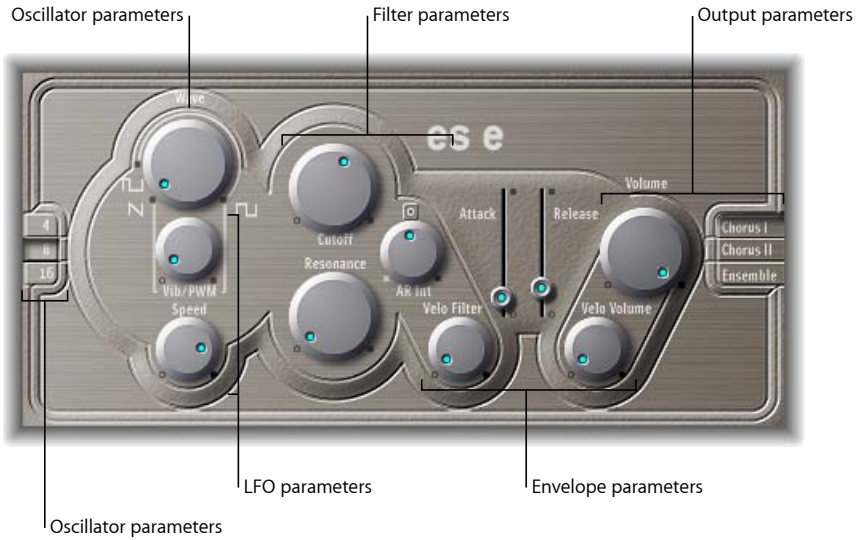
If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics, which will introduce you to the terminology and give you a great overview of different synthesis methods and how they work.

This chapter covers the following:

- Getting to Know the ES E Interface (p. 14)
- Using the ES E Oscillators (p. 15)
- Using the ES E LFO (p. 15)
- Using the ES E Filter (p. 16)
- Using the ES E Envelope (p. 17)
- Using the ES E Output Parameters (p. 17)
- Extended ES E Parameters (p. 18)

Getting to Know the ES E Interface

Before you take a look at the individual parameters of the ES E, this section will familiarize you with the various elements in the ES E graphical interface.



- *Oscillator parameters:* The oscillator Wave and Octave parameters are shown in the area to the left. The oscillator generates the waveforms that form the basis of your sound. See Using the ES E Oscillators.
- *LFO parameters:* The LFO parameters (below the oscillator Wave parameter knob) are used to modulate the sound. See Using the ES E LFO.
- *Filter parameters:* The section to the right of the Oscillator parameters includes the Cutoff (Frequency) and Resonance knobs. You use the filter to contour the waveforms sent from the oscillator. See Using the ES E Filter.
- *Envelope parameters:* The area to the right of the Filter parameters contains the Envelope parameters, which control the level of the sound over time. See Using the ES E Envelope.
- *Output parameters:* The area at the extreme right houses the Volume knob, responsible for the main output level, and the Effect parameters. The effects can be used to color or thicken the sound. See Using the ES E Output Parameters.
- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES E Parameters.

Using the ES E Oscillators

The synthesizer oscillator generates a waveform, which is then sent to other portions of the synthesizer engine for processing or manipulation.

- *Wave knob:* Selects the waveform of the oscillator, which is responsible for the basic color of the tone. The leftmost setting of the Wave parameter causes the oscillators to output sawtooth signals. Across the remaining range, the oscillators output pulse waves, with the average pulse width being defined by the Wave parameter position.
- *4, 8, 16 buttons:* Allow you to switch the pitch in octaves (transpose it up or down). The lowest setting is 16 feet, and the highest is 4 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer (and wider) the pipe, the deeper the tone.

Using the ES E LFO

The LFO (low frequency oscillator) generates a cyclic waveform that is used to modulate the ES E waveform. The behavior and effect of the LFO depend on whether a sawtooth or pulse wave is selected.

- If Wave is set to sawtooth, the LFO modulates the frequency of the waveform, resulting in a vibrato or siren effect—depending on the LFO speed and intensity.
- If Wave is set to a pulse wave, the LFO modulates the waveform's pulse width (PWM).

- *Vib/PWM knob:* Defines the intensity of the LFO modulation.
- *Speed knob:* Sets the frequency of the LFO modulation.

Note: When the pulse width becomes very narrow, the signal sounds as if it is being interrupted ("breaking up"). Given this potential artifact, set the PWM intensity with care, and select the Wave parameter's 12 o'clock position (50% rectangular) for the pulse width, if you want to achieve the maximum modulation range.

Using the ES E Filter

The ES E includes a lowpass filter that lets you contour the output from the oscillator.

- *Cutoff knob:* Controls the cutoff frequency of the ES E filter.
- *Resonance knob:* Boosts or cuts portions of the signal that surround the frequency defined by the Cutoff parameter.

  Note: Increasing the Resonance value results in a rejection of bass (low frequency energy) when using lowpass filters.
- *AR Int knob:* The ES E features one simple envelope generator per voice-offering an Attack and a Release parameter (see Using the ES E Envelope). The AR Int knob defines the amount (depth) of cutoff frequency modulation applied by the envelope generator.
- *Velo Filter knob:* Sets the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator.

  Note: This parameter has no effect if AR Int is set to 0.

Using the ES E Envelope

The AR (Attack and Release) envelope affects both the filter cutoff (AR Int) and the level of the sound over time.

- *Attack slider:* Determines the time that it takes for the signal to reach the initial, desired signal level (the *sustain* level).
- *Release slider:* Determines the time that it takes for the signal to fall from the sustain level to a level of zero.

Using the ES E Output Parameters

The ES E output stage consists of the Volume section and the Chorus/Ensemble buttons.

- *Volume knob:* Sets the overall output level of the ES E.
- *Velo Volume knob:* Sets the amount (depth) of velocity sensitivity to incoming MIDI note events. When it is set to higher values, each note is louder, if struck more firmly. At lower values, the dynamic response is reduced, so that there is little difference when you play a note pianissimo (soft) or forte (loud/hard).

- *Chorus I, Chorus II, and Ensemble buttons:* Click to switch any of these effect variations on or off.
  - Chorus I and Chorus II are typical chorus effects.
  - Chorus II is characterized by a stronger modulation.
  - The Ensemble effect employs a more complex modulation routing, creating a fuller and richer sound.
  - If none of the buttons is active, the effects processor is turned off.

Extended ES E Parameters

The ES E offers three additional parameters that are accessed by clicking the disclosure triangle at the lower left of the interface.

- *Pos. Bender Range:* Changes the positive (upwards) pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES E pitch.
- *Neg. Bender Range:* The default Neg. Bender Range value is Pos PB (Positive Pitch Bend). In essence, this means that only positive pitch bend is available. You can adjust the negative (downwards) pitch bend range in semitone steps-up to 2 octaves (a value of 24).
- *Tune:* Tunes the entire instrument in cents. A cent is 1/100th of a semitone.

ES M

2

The monophonic ES M (ES Mono) synthesizer is a good starting point if you're looking for bass sounds that punch through your mix.

ES-M features an automatic fingered portamento mode, making bass slides easy. It also provides an automatic filter compensation circuit that delivers rich, creamy basses, even when using higher resonance values.

ES M produces sounds using subtractive synthesis. It features an oscillator that generates harmonically rich waveforms. You *subtract* (cut, or filter out) portions of these waveforms, and reshape them, to create new sounds.

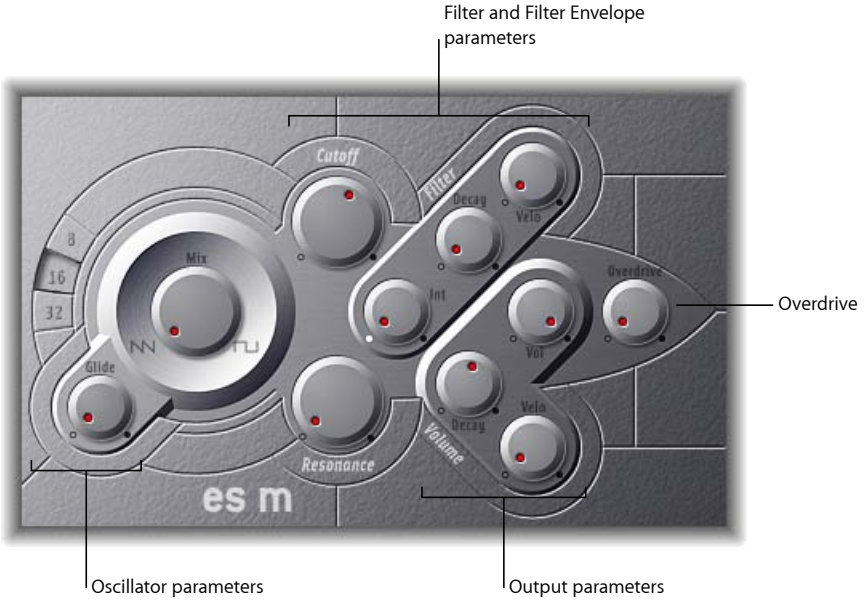
If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics, which will introduce you to the terminology and give you a great overview of different synthesis methods and how they work.

This chapter covers the following:

- Getting to Know the ES M Interface (p. 20)
- Using the ES M Oscillator (p. 21)
- Using the ES M Filter and Filter Envelope (p. 22)
- Using the ES M Level Envelope and Output Controls (p. 23)
- Extended ES M Parameters (p. 23)

Getting to Know the ES M Interface

Before you take a look at the individual parameters of the ES M, this section will familiarize you with the various elements in the ES M graphical interface.



- *Oscillator parameters:* The oscillator Mix and Octave parameters are shown in the area to the left. The oscillator generates the basic waveforms that form the basis of your sound. See Using the ES M Oscillator.
- *Filter and Filter Envelope parameters:* The section to the right of the Oscillator parameters includes the (Cutoff) Frequency and Resonance knobs. The filter is used to contour the waveforms sent from the oscillators. The Filter Envelope parameters are found towards the upper right. These control the filter cutoff over time. See Using the ES M Filter and Filter Envelope.
- *Output parameters:* The angle-shaped area to the lower right contains the Level Envelope and Output parameters, which control the level of the sound over time. The Overdrive knob is located near the right edge of the interface, halfway up. The Overdrive can be used to color or add bite to the sound. See Using the ES M Level Envelope and Output Controls.
- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES M Parameters.

Using the ES M Oscillator

The synthesizer oscillator is used to generate a waveform, which is then sent to other portions of the synthesizer engine for processing or manipulation.

- *Mix knob:* Sets the waveform of the oscillator, which is responsible for the basic color of the tone.
  - Setting the Wave parameter all the way to the left causes the oscillators to output sawtooth signals.
  - Setting the Wave parameter all the way to the right outputs a 50% rectangular wave—which is heard one octave below the sawtooth.
  - For any Wave setting between these extreme positions, the oscillator outputs a crossfaded mix of the two waveforms.
- *8, 16, 32 buttons:* Allow you to switch the pitch in octaves (transpose it up or down). The lowest setting is 32 feet, and the highest is 8 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer (and wider) the pipe, the deeper the tone.
- *Glide knob:* Determines the speed of the glide (the time it takes to slide between note pitches). At a value of 0, no glide effect occurs.

  Note: The ES M always works in a fingered portamento mode, with notes played in a legato style resulting in a glide (portamento) from pitch to pitch.

Using the ES M Filter and Filter Envelope

The ES M includes a lowpass filter that lets you contour the output from the oscillator. The filter features a dedicated envelope.

- *Cutoff knob:* Controls the cutoff frequency of the ES M filter. Its slope is 24 dB/octave.
- *Resonance knob:* Boosts or cuts portions of the signal that surround the frequency defined by the Cutoff parameter.

  Note: Increasing the Resonance value results in a rejection of bass (low frequency energy) when using lowpass filters. The ES M compensates for this side-effect internally, resulting in a more bassy sound.
- *Int knob:* Defines the amount (the intensity or depth) of cutoff frequency modulation applied by the envelope generator.
- *Decay knob:* Defines the decay time of the filter envelope.
- *Velo knob:* Sets the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator.

  Note: The Decay and Velo parameters have no effect if Int is set to 0.

Using the ES M Level Envelope and Output Controls

The output stage of the ES M offers the following parameters.

- *Decay knob:* Sets the decay time of the dynamic stage. The attack, release, and sustain times of the synthesizer are internally set to 0.
- *Velo knob:* Determines the velocity sensitivity of the dynamic stage.
- *Vol knob:* Sets the master output level of the ES M.
- *Overdrive knob:* Controls the level of the integrated overdrive effect.

  Important: To avoid hurting your ears or damaging your speakers, consider turning down the Volume level before setting Overdrive to a high value; then turn it up gradually.

Extended ES M Parameters

The ES M offers three additional parameters that are accessed by clicking the disclosure triangle at the lower left of the interface.

- *Pos. Bender Range:* Changes the positive (upwards) pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES P pitch.
- *Neg. Bender Range:* The default Neg. Bender Range value is Pos PB (Positive Pitch Bend). In essence, this means that only positive pitch bend is available. You can adjust the negative (downwards) pitch bend range in semitone steps-up to 2 octaves (a value of 24).
- *Tune:* Tunes the entire instrument in cents. A cent is 1/100th of a semitone.

ES P

3

The 8-voice ES P (ES Poly) emulates classic polyphonic synthesizers of the 1980s.

It is a versatile instrument that is capable of producing a huge variety of useful musical sounds. The creation of classic analog synthesizer brass sounds is just one of its many strengths.

ES P produces sounds using subtractive synthesis. It features an oscillator that generates harmonically rich waveforms. You *subtract* (cut, or filter out) portions of these waveforms, and reshape them, to create new sounds.

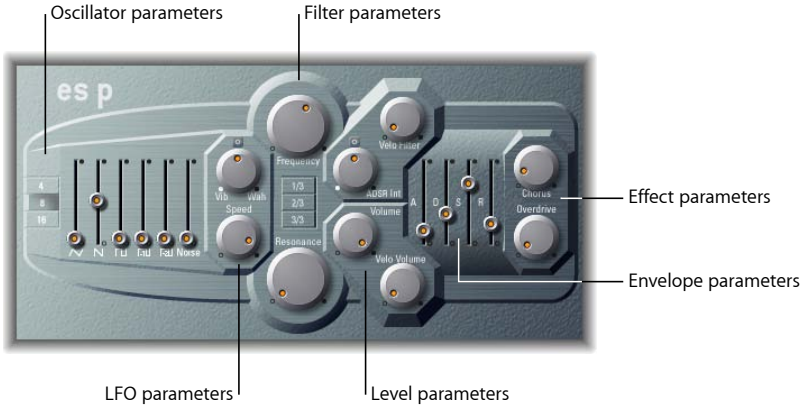
If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics, which will introduce you to the terminology and give you a great overview of different synthesis methods and how they work.

This chapter covers the following:

- Getting to Know the ES P Interface (p. 26)
- Using the ES P Oscillators (p. 27)
- Using the ES P LFO (p. 27)
- Using the ES P Filter (p. 28)
- Using the ES P Envelope and Level Controls (p. 29)
- Using the Integrated ES P Effects (p. 30)
- Extended ES P Parameters (p. 30)

Getting to Know the ES P Interface

Before you take a look at the individual parameters of the ES P, this section will familiarize you with the various elements in the ES P graphical interface.



- *Oscillator parameters:* The Oscillator sliders are shown in the area to the left. The Octave parameters are also found in this section. The oscillators generate the basic waveforms that form the basis of your sound. See Using the ES P Oscillators.
- *LFO parameters:* The LFO parameters (to the right of the Oscillator parameters) are used to modulate the sound. See Using the ES P LFO.
- *Filter parameters:* The vertical column in the center includes the (Cutoff) Frequency and Resonance knobs and the Key Follow buttons. The filter is used to contour the waveforms sent from the oscillators. See Using the ES P LFO.
- *Envelope and Level parameters:* The area to the right of the Filter parameters contains the Envelope and Level parameters, which control the level of the sound over time. See Using the ES P Envelope and Level Controls.
- *Effect parameters:* The area at the extreme right contains the Chorus and Overdrive parameters. These can be used to color or thicken the sound. See Using the Integrated ES P Effects.
- *Extended parameters:* Not shown in the image, the extended parameters are accessed by clicking the triangle at the lower left of the interface. These parameters include bend and tuning functions. See Extended ES P Parameters.

Using the ES P Oscillators

The ES P features several oscillators that output different waveforms. These signals can be mixed together—at different levels—providing countless variations of the raw material used for your sounds.

- *Oscillator sliders:* Set the level of the waveforms output by the oscillators.
  - In addition to triangular, sawtooth, and rectangular waves, the rectangular waves of two sub-oscillators are also available. The left sub-oscillator fader is one octave lower than the main oscillators, and the right sub-oscillator fader is two octaves lower. Use these to fatten up the sound.
  - The pulse width of all rectangular waves is fixed at 50%.
  - The rightmost fader adds white noise to the mix. This is the raw material for classic synthesizer sound effects, such as ocean waves, wind, and helicopters.
- *4, 8, 16 buttons:* Allow you to switch the pitch in octaves (transpose it up or down). The lowest setting is 16 feet, and the highest is 4 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer (and wider) the pipe, the deeper the tone.

Using the ES P LFO

The ES P features an LFO (low frequency oscillator), which can do either of the following:

- Modulate the frequency of the oscillators, resulting in a vibrato.

- Modulate the cutoff frequency of the dynamic lowpass filter, resulting in a wah wah effect.

- *Vib/Wah knob:* Turn to the left to set a vibrato; turn to the right to cyclically modulate the filter.
- *Speed knob:* Sets the rate of the vibrato or cutoff frequency modulation.

Using the ES P Filter

The ES P includes a lowpass filter that lets you contour the output from the oscillator.

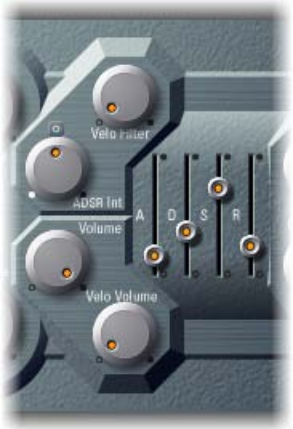
- *Frequency knob:* Controls the cutoff frequency of the ES P's lowpass filter.
- *Resonance knob:* Boosts/cuts portions of the signal that surround the frequency defined by the Frequency knob.

Note: Increasing the Resonance value results in a rejection of bass (low frequency energy) when using lowpass filters. The ES P compensates for this side-effect internally, resulting in a more bassy sound.

- *1/3, 2/3, 3/3 (Key Follow) buttons:* The cutoff frequency can be modulated by MIDI note number (keyboard position); you may know this parameter as *keyboard follow* on other synthesizers. Enable one of the 1/3, 2/3, 3/3 buttons to choose one-third, two-thirds, or full-keyboard follow. If no button is active, the key you strike won't affect the cutoff frequency. This makes the lower notes sound relatively brighter than the higher ones. If 3/3 is chosen, the filter follows the pitch, resulting in a constant relationship between cutoff frequency and pitch. This is typical of many acoustic instruments where higher notes sound both brighter in tone and higher in pitch.
- *ADSR Int knob:* Defines the amount (depth) of cutoff frequency modulation applied by the envelope generator (see Using the ES P Envelope and Level Controls).
- *Velo Filter knob:* Sets the velocity sensitivity of the cutoff frequency modulation applied by the envelope generator. The main envelope generator (ADSR) modulates the cutoff frequency over the duration of a note. The intensity of this modulation can respond to velocity information. If you play pianissimo (velocity = 1), the modulation is minimal. If you strike with the hardest fortissimo (velocity = 127), the modulation is more intense.

Using the ES P Envelope and Level Controls

The ES P features an ADSR envelope that affects both the filter cutoff (ADSR Int) and the level of the sound over time. This section also covers the master level control parameters.



- *Attack slider:* Determines the time it takes for the signal to reach the initial, desired signal level (the *sustain level*).
- *Decay slider:* Determines the time it takes for the signal to fall from the attack level to the sustain level.

- *Sustain slider:* Determines the desired signal level (the *sustain* level).
- *Release slider:* Determines the time it takes for the signal to fall from the sustain level to a level of zero.
- *Volume knob:* Sets the overall output level of the ES P.
- *Velo Volume knob:* Sets the amount (depth) of velocity sensitivity to incoming MIDI note events. When set to higher values, each note is louder if struck more firmly. At lower values, the dynamic response is reduced, so that there is little difference when you play a note pianissimo (soft) or forte (loud/hard).

Using the Integrated ES P Effects

The ES P offers integrated stereo chorus and overdrive effects. These are based on similar effects processors found in the affordable Japanese synthesizers of the 1980s—which the ES P, itself, emulates.

- *Chorus knob:* Sets the intensity (depth) of the integrated chorus effect.
- *Overdrive knob:* Determines the overdrive/distortion level of the ES P output.

  Important: To avoid hurting your ears or damaging your speakers, consider turning down the Volume level before setting Overdrive to a high value; then turn it up gradually.

Extended ES P Parameters

The ES P offers three additional parameters that are accessed by clicking the disclosure triangle at the lower left of the interface.

- *Pos. Bender Range:* Changes the positive (upwards) pitch bend range in semitone steps. This allows you to use the pitch bend controller of your keyboard to bend the ES P pitch.

- *Neg. Bender Range:* The default Neg. Bender Range value is Pos PB (Positive Pitch Bend). In essence, this means that only positive pitch bend is available. You can adjust the negative (downwards) pitch bend range in semitone steps-up to 2 octaves (a value of 24).
- *Tune field:* Tunes the entire instrument in cents. A cent is 1/100th of a semitone.

ES1

4

The ES1 emulates the circuits of analog synthesizers in a simple, streamlined interface.

ES1 produces sounds using subtractive synthesis. It features an oscillator and sub-oscillator that generate harmonically rich waveforms. You *subtract* (cut, or filter out) portions of these waveforms and reshape them to create new sounds. The ES1's tone-generation system also provides flexible modulation options that make it easy to create punchy basses, atmospheric pads, biting leads, and sharp percussion.

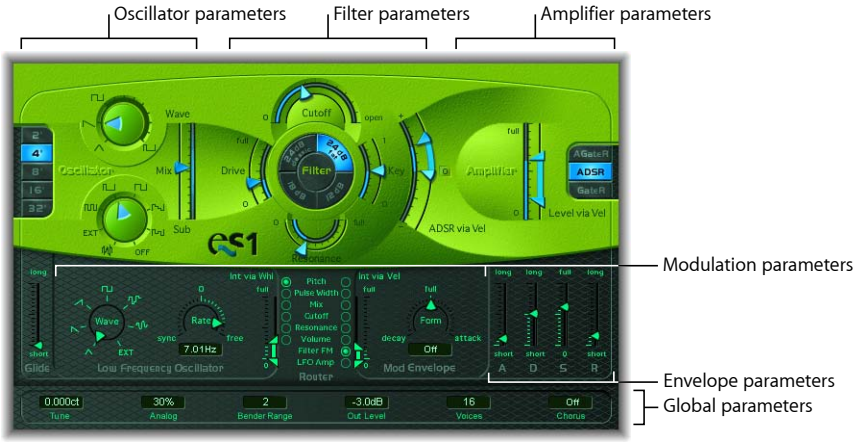
If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics, which will introduce you to the terminology and give you a great overview of different synthesis systems and how they work.

This chapter covers the following:

- Getting to Know the ES1 Interface (p. 34)
- Using the ES1 Oscillators (p. 35)
- Using the ES1 Filter Parameters (p. 37)
- Using the ES1 Amplifier Parameters (p. 38)
- Using the ES1 Envelope Parameters (p. 39)
- Modulating the Sound of the ES1 (p. 41)
- Adjusting Global ES1 Parameters (p. 44)
- ES1 MIDI Controller List (p. 45)

Getting to Know the ES1 Interface

Before you take a look at the individual parameters of the ES1, this section will familiarize your with the various elements that constitute the ES1's graphical interface, which is broken down into six main areas.



- *Oscillator parameters:* Shown to the left, the oscillators generate the basic waveforms that form the basis of your sound. See Using the ES1 Oscillators. A number of related parameters that directly influence the overall sound, such as Tune, are found in the Global Parameters section of the interface. See Adjusting Global ES1 Parameters.
- *Filter parameters:* These include the circular Filter area, and Drive and Key scaling parameters. You use the filter to contour the waveforms sent from the oscillators. See Using the ES1 Filter Parameters.
- *Amplifier parameters:* The area to the right contains the Amplifier parameters. See Using the ES1 Amplifier Parameters.
- *Envelope parameters:* The ADSR sliders in the lower-right corner of the ES1 can be used to control both filter cutoff and the amplifier level over time. See Using the ES1 Envelope Parameters.
- *Modulation parameters:* The dark green/gray area houses the modulation sources, modulation router, modulation envelope, and amplitude envelope, which you can use to modulate the sound in a number of ways. See Modulating the Sound of the ES1.
- *Global parameters:* Global sound control parameters are located on the bottom green/gray strip. This is where you can assign and adjust global tuning, activate the in-built chorus, and more. The chorus can be used to color or thicken the sound. See Adjusting Global ES1 Parameters.

Using the ES1 Oscillators

The ES1 features a primary oscillator and a sub-oscillator. The primary oscillator generates a waveform, which is then sent to other parts of the synthesizer for processing or manipulation. The sub-oscillator generates a secondary waveform one or two octaves below that of the primary oscillator.

- *Wave knob:* Selects the waveform of the primary oscillator, which is responsible for the basic color of the tone. See Setting ES1 Oscillator Waveforms.
- *Mix slider:* Defines the level relationship between the primary and sub-oscillator signals. When the sub-oscillator is switched off, its output is completely removed from the signal path.
- *Sub knob:* The sub-oscillator generates square, pulse, and white-noise waveforms. It also allows you to route a side chain signal through the ES1 synthesizer engine (see Using the ES1 Sub-Oscillator).
- *2, 4, 8, 16, 32 buttons:* Transpose the pitch of the oscillators up or down by octaves. The lowest setting is 32 feet, and the highest is 2 feet. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer and wider the pipe, the deeper the tone (for a description of the global Tune parameters, see Adjusting Global ES1 Parameters).

Setting ES1 Oscillator Waveforms

The following table outlines how the oscillator waveform affects your synthesizer sound.

Waveform	Basic tone	Comments
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes, pads
Square	Hollow and “woody” sounding	Useful for basses, clarinets, and oboes

Waveform	Basic tone	Comments
Pulse	“Nasal” sounding	Great for reed instruments, synth blips, basses

Using Pulse Width Modulation in the ES1

You can freely set any pulse width in-between the square wave and pulse wave symbols of the Wave knob. The pulse width can also be automatically modulated in the modulation section (see Using the ES1 Router). Modulating the pulse width with a slowly cycling LFO, for example, allows periodically mutating, fat bass sounds.

Using the ES1 Sub-Oscillator

The sub-oscillator provides the following waveform options:

- A square wave that plays one or two octaves below the frequency of the primary oscillator.
- A pulse wave that plays two octaves below the frequency of the primary oscillator.
- Variations of these waveforms with different mixes and phase relationships. Their use results in different sounds.
- White noise which is useful for creating percussion sounds, as well as wind, surf, and rain sounds.
- You can also completely disable the sub-oscillator by choosing OFF.

Processing Side Chain Signals Through the ES1

The sub-oscillator of the ES1 allows you to run an external channel strip signal through the ES1 synthesizer engine, by using a side chain.

To process a channel strip signal through the ES1 synthesizer engine

1. Set the Sub knob to EXT.
2. Choose the side chain source channel strip from the Side Chain menu at the top of the plug-in window.

Using the ES1 Filter Parameters

This section outlines the filter parameters of the ES1.

- *Cutoff slider:* Controls the cutoff frequency of the ES1's lowpass filter.
- *Resonance slider:* Cuts or boosts the portions of the signal that surround the frequency defined by the Cutoff parameter. Boost can be set so intensively that the filter begins to oscillate by itself (see Driving the ES1 Filter to Self-Oscillation).

  Tip: You can simultaneously adjust the cutoff frequency and resonance parameters by dragging vertically (cutoff) or horizontally (resonance) on the word *Filter*, found between the Slope buttons.
- *Slope buttons:* The lowpass filter offers four different slopes of band rejection above the cutoff frequency. The four settings are arranged clockwise from the top left, as follows:
  - *24 dB classic:* Mimics the behavior of a Moog filter. Turning up the resonance results in a reduction of the low end of the signal.
  - *24 dB fat:* Compensates for the reduction of low frequency content caused by high Resonance values. This resembles the behavior of an Oberheim filter.
  - *12 dB:* Provides a soft, smooth sound which is reminiscent of the early Oberheim SEM.
  - *18 dB:* Tends to resemble the filter sound of Roland's TB-303.
- *Drive slider:* An input level control, which allows you to overdrive the filter. Use it to change the behavior of the Resonance parameter, which will eventually distort the sound of the waveform.
- *Key slider:* Controls how strongly the keyboard pitch (the note number) modulates the filter's cutoff frequency:
  - If Key is set to zero, the cutoff frequency won't change, no matter which key you strike. This makes the lower notes sound relatively brighter than the higher ones.

- If Key is set to maximum, the filter follows the pitch, resulting in a constant relationship between cutoff frequency and pitch. This mirrors the properties of many acoustic instruments where higher notes sound both brighter in tone and higher in pitch.

- *ADSR via Vel slider:* Determines how note velocity affects the filter cutoff frequency modulation—caused by the envelope generator (see Using the ES1 Envelope Parameters).

Driving the ES1 Filter to Self-Oscillation

You can make the ES1 filter output a sine wave by following the steps below. This will allow you to "play" the filter-generated sine wave with the keyboard.

To output a sine wave from the filter

1. Switch the Sub knob to OFF.
2. Move the Mix slider to the very bottom (Sub).
3. Turn the Resonance knob to the maximum position.

Using the ES1 Amplifier Parameters

The parameters in the ES1 Amplifier section allow you to fine-tune the behavior of your sound's level. These are separate from the global Out Level parameter, found on the strip at the bottom of the interface, and which acts as the ES 1's master volume (see Adjusting Global ES1 Parameters).

- *Level via Vel slider:* Determines how note velocity affects the synthesizer level-set by the global Out Level parameter. The slider's upper arrow indicates the level when you play fortissimo. The lower arrow indicates the level when you play pianissimo (velocity = 1). The greater the distance between the arrows (indicated by the blue bar), the more the volume is affected by incoming velocity messages. You can simultaneously adjust the modulation range and intensity by dragging the blue bar-between the arrows-and moving both arrows at once.

- *Amplifier Envelope Selector buttons:* The AGateR, ADSR, and GateR buttons define which of the ADSR envelope generator controls have an effect on the amplifier envelope (see Using the Envelope to Control the ES1 Amplifier).

Using the ES1 Envelope Parameters

The ES1 features an attack, decay, sustain, and release (ADSR) envelope that can shape the filter cutoff and the level of the sound over time.

- *Attack slider:* Determines the amount of time that it takes for the envelope to reach the initial, desired level.
- *Decay slider:* Determines how long it takes for the envelope to fall to the sustain level, following the initial attack time.
- *Sustain slider:* Sets the sustain level, which is held until the keyboard key is released.
- *Release slider:* Determines the length of time that it takes the envelope to fall from the sustain level to a level of zero.

Using the Envelope to Control ES1 Cutoff Frequency

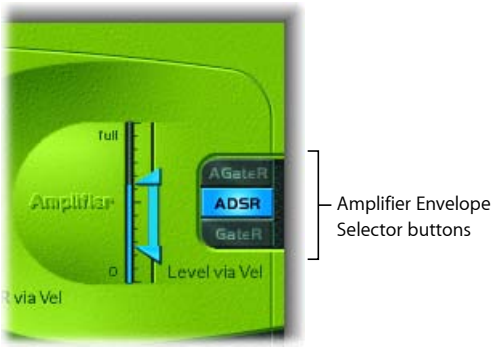
The envelope generator modulates the filter cutoff frequency over the duration of a note. The intensity of this modulation—and how it responds to velocity information—is set by the ADSR via Vel slider arrows (in the Filter Parameters section).

The modulation range is determined by the two arrows. The minimum amount of modulation is indicated by the lower arrow. The upper arrow indicates the maximum amount of modulation. The blue bar between the arrows shows the dynamic range of this modulation. You can simultaneously adjust the modulation range and intensity by dragging the blue bar and moving both arrows at once.

Tip: If you're unfamiliar with these parameters, set the Cutoff to a low value, Resonance to a high value, and move both of the ADSR via Vel arrows upwards. Constantly strike a note on the keyboard while changing the arrows to discover how these parameters work.

Using the Envelope to Control the ES1 Amplifier

The AGateR, ADSR, and GateR buttons (in the Amplifier Parameter section) define which of the ADSR envelope generator controls affect the amplifier envelope. All ADSR parameters remain active for the filter.



The letters *A, D, S,* and *R* refer to the attack, decay, sustain, and release phases of the envelope (see Using the ES1 Envelope Parameters). *Gate* refers to a control signal used in analog synthesizers that tells an envelope generator that a key is pressed. As long as an analog synthesizer's key is pressed, the gate signal maintains a constant voltage. When Gate is used as a modulation source in the voltage-controlled amplifier (instead of the envelope itself), it creates an organ type envelope without any attack, decay, or release-an even, sustained sound.

In the ES1, the Amplifier Envelope selector buttons have the following effect on played notes:

- *AGateR:* The Attack and Release sliders of the ADSR Envelope control the attack and release phases of the sound. In-between these phases, the Gate control signal is used to maintain a constant level, while the note is held. As soon as you release the key, the release phase begins. The Decay and Sustain sliders of the ADSR Envelope have no impact on the sound's level.
- *ADSR:* This is the standard operating mode of most synthesizers, where the level of the sound over time is completely controlled by the ADSR Envelope.

- *GateR:* The Gate control signal is used to maintain a constant level, while the note is held. As soon as you release the key, the release phase begins. The Attack, Decay and Sustain sliders of the ADSR Envelope have no impact on the sound's level.

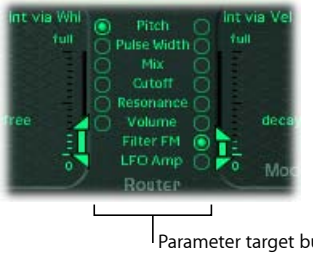
Modulating the Sound of the ES1

The ES1 offers a number of simple, yet flexible, modulation routing options. Modulation is used to add animation to your sound over time, making it more interesting, lively, or realistic. A good example of this type of sonic animation is the vibrato used by orchestral string players.

- *LFO parameters:* Used to modulate other ES1 parameters. See Using the ES1 LFO.
- *Router:* Allows you to choose which ES1 parameters are modulated. See Using the ES1 Router below.
- *Modulation Envelope:* A dedicated modulation control source, it can directly control other ES1 parameters, or it can control the LFO level. See Using the ES1 Mod Envelope.

Using the ES1 Router

The router is used to determine which of the ES1 parameters are modulated by the LFO (see Using the ES1 LFO) and/or by the Modulation Envelope (see Using the ES1 Mod Envelope). The parameter target buttons in the left column enable LFO modulation, and the ones in the right column set the target for the Modulation Envelope.



- *Pitch buttons:* Turn on to modulate the pitch—the frequency—of the oscillators.
- *Pulse Width buttons:* Turn on to modulate the pulse width of the pulse wave.

- *Mix buttons:* Turn on to modulate the mix between the primary and sub-oscillators.
- *Cutoff buttons:* Turn on to modulate the cutoff frequency of the filter.
- *Resonance buttons:* Turn on to modulate the resonance of the filter.
- *Volume buttons:* Turn on to modulate the main volume.
- *Filter FM button (modulation envelope only):* Turn on to use the triangle wave of the oscillator to modulate the filter cutoff frequency. This modulation can result in a pseudo-distortion of the sound, or it can create metallic, FM-style sounds. The latter occurs when the only signal you can hear is the self-oscillation of the resonating filter (see Driving the ES1 Filter to Self-Oscillation).
- *LFO Amp (modulation envelope only):* Turn on to modulate the overall amount of LFO modulation.

Using the ES1 LFO

The LFO (low frequency oscillator) generates an adjustable, cyclic waveform that can be used to modulate other ES1 parameters (as discussed in Using the ES1 Router).

- *Wave knob:* Sets the LFO waveform. You can choose from triangle, ascending and descending sawtooth, square wave, sample & hold (random), and a lagged, smoothly changing random wave. Each of these waveforms cycles in its own way, providing different types of modulation. You can also assign a side chain signal as a modulation source (EXT). Choose the side chain source channel strip from the Side Chain menu at the top of the plug-in window.
- *Rate slider and field:* Defines the speed—the frequency—of the LFO waveform cycles.
  - If you set values to the left of zero, the LFO phase is locked to the tempo of the host application—with phase lengths adjustable between 1/96 bar and 32 bars. If you select values to the right of zero, the LFO phase runs freely.
  - When set to zero, the LFO outputs at a constant and full level, which allows you to manually control the LFO speed with your keyboard's modulation wheel. This could be useful, for example, if you wanted to change the pulse width by moving your keyboard's modulation wheel. The pulse width would be chosen as the LFO modulation target (left router column), and the modulation intensity range would be determined by the Int via Whl setting.

- *Int via Whl slider:* The upper arrow defines the intensity of the LFO modulation if the modulation wheel (MIDI controller 1) is set to its maximum value. The lower arrow defines the amount of LFO modulation if the modulation wheel is set to zero. The distance between the arrows—shown by a green bar—indicates the range of your keyboard's modulation wheel. You can simultaneously adjust the modulation range and intensity by dragging the green bar, thus moving both arrows at once. Note that as you do so, the arrows retain their relative distance from each other.

Using the ES1 Mod Envelope

The modulation envelope allows you to set a percussive type of decay envelope by choosing low values, or attack type envelopes with high values.

- *Form slider and field:* Allows you to fade in (attack) or fade out (decay) the modulation. When set to the full position, the modulation envelope is turned off.
- *Int via Vel slider:* The top arrow controls the upper modulation intensity setting for the modulation envelope—if you strike a key with the hardest fortissimo (velocity = 127). The bottom arrow controls the lower modulation intensity setting for the modulation envelope—if you strike a key with the softest pianissimo (velocity = 1). The green bar between the arrows displays the impact of velocity sensitivity on the intensity of the modulation envelope. You can simultaneously adjust the modulation range and intensity by dragging the green bar, thus moving both arrows at once. Note that as you do so, the arrows retain their relative distance from each other.

Controlling ES1 Parameters and the LFO with the Mod Envelope

The Modulation Envelope can directly modulate the parameter chosen in the router. It essentially determines the time it takes for the modulation to fade in or fade out. At its center position, which can be accessed by clicking the middle mark, the modulation intensity is static—no fade in or fade out will occur. When set to its full value, it delivers a constant level.

To modulate a parameter with velocity

1. Select a modulation destination (Pulse Width, for example).
2. Set Form to full, and adjust the Int via Vel parameter as needed.

   This results in a velocity-sensitive modulation of the oscillator pulse width.

   More interestingly, the Modulation Envelope can directly control the LFO level when you click the LFO Amplitude button in the right column of the router.

To fade the LFO modulation in or out

- Choose a positive Form value (towards attack) to *fade in* the LFO modulation. The higher the value, the longer it takes for you to hear the modulation.
- Choose a negative value (towards decay) to *fade out* the LFO modulation. The lower the value (closer to full), the shorter the fade out time is.

LFO control with envelopes is most often used for delayed vibrato, which is a technique many instrumentalists and singers employ to intonate longer notes.

To set up a delayed vibrato

1. Position the Form parameter toward the right (attack).
2. Choose Pitch as the LFO target (left column).
3. Choose the triangular wave as the LFO waveform.
4. Select an LFO Rate of about 5 Hz.
5. Set the upper Int via Wheel arrow to a low value, and the lower arrow to 0.

Adjusting Global ES1 Parameters

This section covers global parameters that affect the overall sound, or behavior, of the ES1. The global parameters are found in the strip that spans the lower edge of the ES1 interface. The Glide slider is displayed above the left end of the strip.

- *Glide slider:* Defines the amount of time it takes to slide between the pitches of each triggered note. The Glide trigger behavior depends on the value chosen for the Voices parameter (see below).
- *Tune field:* Tunes the entire instrument in cents. A cent is 1/100th of a semitone.
- *Analog field:* Slightly alters the pitch of each note, and the cutoff frequency, in a random fashion. This emulates the random oscillator detuning and filter fluctuations that can occur in polyphonic analog synthesizers—due to heat and age.
- *Voices field:* Sets the maximum number of notes that can be played simultaneously. Each ES1 instance offers a maximum of 16 voices. Fewer played voices require less CPU power.
- *Bender Range field:* Changes the sensitivity of the pitch bender. Adjustments are in semitone steps.
- *Neg Bender Range slider (Extended Parameters area):* Adjusts the negative (downwards) pitch bend range in semitone steps. The default value is Pos PB (Positive Pitch Bend), which essentially means that there is no downward pitch bend available. Click the small triangle at the lower left of the ES1 interface to access the Extended parameters area.

- *Chorus field:* The ES1 offers two classic stereo chorus effects and one ensemble effect.
  - Off deactivates the in-built chorus circuit.
  - C1 and C2 are typical chorus effects. C2 is a variation of C1 and is characterized by a stronger modulation.
  - Ens (Ensemble) employs a more complex modulation routing, creating a fuller and richer sound.
- *Out Level field:* Controls the master volume of the ES1.

Using the ES1 Analog Parameter

When the Analog parameter is set to a value of zero, the oscillator cycle start points of all triggered voices are synchronized. This may be useful for percussive sounds, when you want to achieve a sharper attack characteristic.

Use of Analog values higher than zero allows the oscillators of all triggered voices to cycle freely. Use higher values if you're aiming for a warm, analog type of sound—where subtle sonic variations occur for each triggered voice.

Using the Voices Parameter in the ES1

When the Voices parameter is set to Legato, the ES1 behaves like a monophonic synthesizer—with single trigger and fingered portamento engaged. This means that if you play legato, a portamento will occur (the portamento time is defined by the Glide slider). If you release each key before pressing a new one, the envelope will not be triggered by the new note, and there will be no portamento.

Tip: You can use this feature to create pitch bend effects—without touching the pitch bender—so make sure you select a higher Glide value when using the Legato setting.

ES1 MIDI Controller List

The ES1 responds to the following MIDI continuous controller numbers (CC).

Controller number	Parameter name
12	Oscillator pitch buttons
13	Oscillator waveform
14	Mix slider
15	Waveform of sub-oscillator
16	Drive slider
17	Cutoff slider
18	Resonance slider
19	Slope buttons
20	ADSR via Vel:lower slider

Controller number	Parameter name
21	ADSR via Vel:upper slider
22	Attack slider
23	Decay slider
24	Sustain slider
25	Release slider
26	Key slider
27	Amplifier Envelope Selector buttons
28	Level via Velocity:lower slider
29	Level via Velocity:upper slider
30	Chorus parameter
31	Modulation envelope target
102	Modulation Envelope form slider
103	Modulation envelope:Int via Vel parameter:lower slider
104	Modulation envelope:Int via Vel parameter:upper slider
105	LFO rate
106	LFO waveform
107	LFO modulation target
108	LFO:Int via Whl:lower slider
109	LFO:Int via Whl:upper slider
110	Glide slider
111	Tune parameter
112	Analog parameter
113	Bender Range parameter
114	Out Level parameter
115	Voices parameter

ES2

5

The ES2 synthesizer combines a powerful tone generation system with extensive modulation features. It seamlessly blends subtractive synthesis and elements of FM and wavetable synthesis methods to generate an extraordinary variety of sounds. This makes it the perfect choice for creating powerful pads, evolving textures, rich basses, or synthetic brass.

If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics, which will introduce you to the fundamentals and terminology of different synthesis systems.

This chapter covers the following:

- ES2 Architecture and Features (p. 48)
- Getting to Know the ES2 interface (p. 49)
- Using the ES2 Oscillators (p. 50)
- Using the ES2's Global Parameters (p. 62)
- Using the ES2 Filters (p. 64)
- Using the ES2 Amplifier Parameters (p. 74)
- Working with Modulation in the ES2 (p. 76)
- Using the ES2's Integrated Effect Processing Section (p. 106)
- Creating Random ES2 Sound Variations (p. 108)
- Using Macro Controls and Assigning Controllers in the ES2 (p. 110)
- ES2 Tutorial: Creating Sounds (p. 112)
- ES2 Tutorial: Using Templates to Create Sounds (p. 123)

ES2 Architecture and Features

The three oscillators of the ES2 provide classic analog synthesizer waveforms (including noise) and 100 single-cycle waveforms, known as Digiwaves. This raw material forms the basis for sounds that range from fat analog to harsh digital sounds, or hybrids of the two. You can also cross-modulate oscillators, making it easy to create FM-style sounds. Further options include the ability to synchronize and ring-modulate the oscillators, or to mix a sine wave directly into the output stage, to thicken the sound.

ES2 features a flexible modulation router that offers up to ten simultaneous (user-defined) modulation routings. These can be used alongside a number of hard-wired routings. Further modulation options include the unique Planar Pad—which provides control of two parameters on a two-dimensional grid. The Planar Pad itself can be controlled by the sophisticated Vector Envelope. This is a multipoint, loop-capable envelope that makes it easy to create complex, evolving sounds.

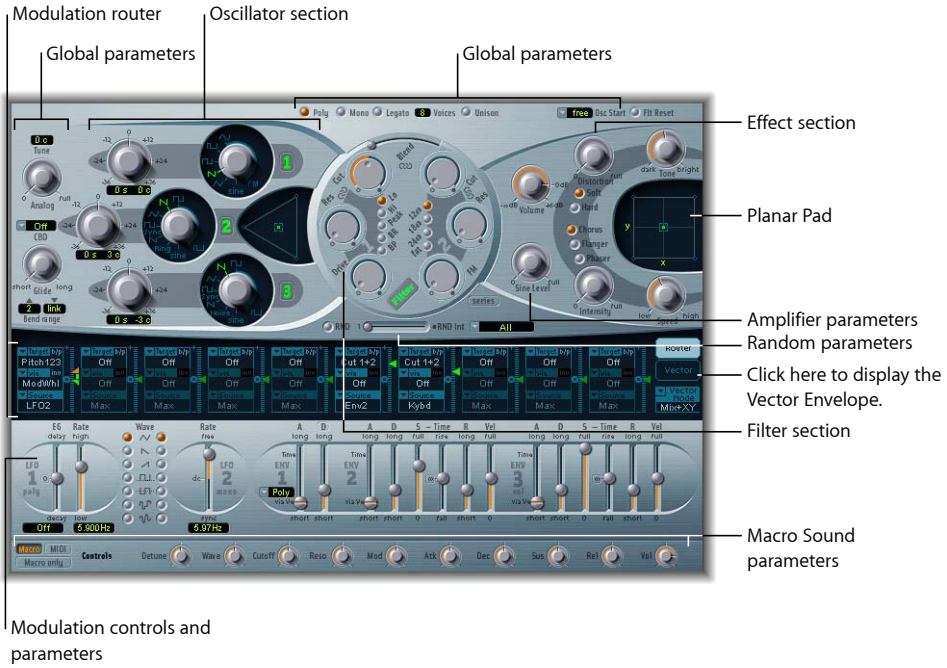
Lastly, Distortion, Chorus, Phaser, and Flanger effects are built into the ES2.

If you'd like to begin experimenting straight away, there are a number of settings that can be loaded which will help you to get started. These are accompanied by two tutorial sections which provide tips and information, and invite you to explore the ES2. See ES2 Tutorial: Creating Sounds and ES2 Tutorial: Using Templates to Create Sounds.

Note: Within the parameter description sections, you will often find information about the use of parameters as modulation targets or sources. This underlines one of the ES2's greatest strengths—namely, the vast modulation possibilities it offers. It is recommended that you read through the chapter, and return to these "modulation information" sections (see below) when, and if, you need to refer to them. Descriptions on the use of all modulation and control options, including reference tables, are covered in Working with Modulation in the ES2.

Getting to Know the ES2 interface

Before you take a look at the individual parameters and functions of the ES2, this section will familiarize you with the various elements of the ES2's graphical interface, which is broken down into the following main areas.



- *Oscillator section:* The oscillator parameters are shown at the upper left of the ES2 interface. The Triangle is used to set the mix relationships between the three oscillators. See Using the ES2 Oscillators.
- *Global parameters:* A number of related global parameters that directly influence the overall output of the ES2—such as Tune—are found to the left of the oscillators, and above the amplifier and filter parameters. See Using the ES2's Global Parameters.
- *Filter section:* The circular Filter area, including the Drive and Filter FM parameters. See Using the ES2 Filters.
- *Amplifier parameters:* The area at the top right contains the output parameters, where you can set the overall volume of the ES2, and add a sine signal at the output stage. See Using the ES2 Amplifier Parameters.

- *Modulation router or Vector Envelope:* The dark strip across the center of the ES2 interface is shared by the modulation router and the Vector Envelope. You can use the buttons at the right end of this section to switch between the two.
  - The router links modulation sources, such as the envelopes and other parameters shown in the lower portion of the interface, to modulation targets, such as the oscillators and filters. See Getting to Know the ES2's Modulation Router.
  - The Vector Envelope is an extremely flexible and powerful envelope generator, which provides extensive control over your sound. See Getting to Know the ES2's Vector Envelope.
- *Modulation controls and parameters:* The area immediately below the router is where you can assign and adjust the modulation generator parameters (such as LFO and envelope parameters). See Working with Modulation in the ES2.
- *Planar Pad:* The square area at the top right is a two-dimensional controller known as the Planar Pad—which facilitates the simultaneous manipulation of two, freely assignable, parameters. The Planar Pad can be controlled using the mouse, another controller, or the Vector Envelope. See Using the ES2's Planar Pad.
- *Effect section:* The built-in effect processing options are found to the right of the Output parameters. See Using the ES2's Integrated Effect Processing Section.
- *Random parameters:* Used to randomize sound parameters, the Random parameters are found below the circular Filter area. See Creating Random ES2 Sound Variations.
- *Macro and MIDI controller parameters:* The area located on the thin, gray strip at the bottom can display either Macro parameters or MIDI controller assignments. The preassigned macro sound parameters are perfect for quick tweaks to the ES2's sound (and that of ES2-based GarageBand instruments). You can reassign MIDI control numbers for these parameters if desired. See Using the ES2 Macro Control Parameters.

Using the ES2 Oscillators

The synthesizer oscillators are used to generate one or more waveforms. This signal is then sent to other portions of the synthesizer engine for shaping, processing, or manipulation.

Before the parameters are discussed, a few of the special features available to you in the ES2 oscillator section should be mentioned.

- Oscillators 2 and 3 are almost identical to each other, but they differ from Oscillator 1.
- Oscillator 1 can be frequency modulated by Oscillator 2, for FM synthesis sounds.
- Oscillators 2 and 3 can be synchronized to, or ring modulated with, Oscillator 1. They also have rectangular waves with either user-defined fixed-pulse widths or pulse width modulation (PWM) features.

- You can use the modulation router to simultaneously change the pulse widths of the rectangular waves of Oscillator 1 and the synchronized and ring-modulated rectangular waves of Oscillators 2 and 3.

Getting to Know the ES2 Oscillators

The following section gives you an overview of the parameters available for each oscillator. You can find the oscillators in the area at the top left of the ES2 interface.

- *Oscillator on/off buttons:* Click the oscillator number to the right of each oscillator to activate or deactivate each oscillator independently. A green numeric button indicates an active oscillator. A gray numeric button denotes an inactive oscillator. Deactivating an oscillator saves computer processing power as you are not merely muting it—you are actually turning it off.
- *Wave knobs:* Sets the type of waveform that an oscillator generates. The waveform is responsible for the basic tonal color. See Using the ES2's Basic Oscillator Waveforms.
- *(Coarse) Frequency knobs:* Sets the oscillator's pitch—in semitone steps, over a range of ±3 octaves. As an octave consists of 12 semitones, the ±12, 24, and 36 settings represent octaves.
- *(Fine) Frequency value field:* Used to fine-tune the oscillator frequency (pitch). The value display works as follows: the left numbers show the semitone setting, the right numbers show the cent (1 cent = 1/100th semitone) setting. These are denoted by an *s* or *c* to the right of the value. You can adjust these two values independently. For example, an oscillator with the value 12 s 30 c sounds an octave (12 semitones) and 30 cents higher than an oscillator with the value 0 s 0 c.

- *Oscillator Mix (Triangle):* Dragging the square icon in the Triangle cross-fades (sets the level relationships) between the three oscillators. See Setting the Oscillator Level Balance in the ES2.

Using the ES2's Basic Oscillator Waveforms

All of the ES2 oscillators output a number of standard waveforms—sine, pulse, rectangular, sawtooth, and triangular waves—or, alternately, any of 100 Digiwaves (see Using Digiwaves in the ES2). The following table includes the basic waveforms:

Waveform	Basic tone	Comments
Pulse/Rectangular	Nasal sounding	Great for reed instruments, synth blips, and basses
Square	Hollow and woody sounding	Useful for basses, clarinets, and oboes. The pulse width of (Oscillator 2 and 3) square waveforms can be smoothly scaled between 50% and the thinnest of pulses.
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes and pad sounds.
Sine	A pure tone	The sine wave of Oscillator 1 can be frequency modulated by Oscillator 2. This kind of modulation forms the basis of FM synthesis (see Using Frequency Modulation in the ES2).

Oscillators 2 and 3 also offer the selection of:

- A rectangular wave, synchronized to Oscillator 1.
- A sawtooth wave, synchronized to Oscillator 1.
- A ring modulator, which is fed by the output of Oscillator 1 and a square wave from Oscillator 2.
- Colored noise for Oscillator 3 (see Using Noise in the ES2 (Oscillator 3 Only)).

Oscillator synchronization and ring modulation enable the creation of very complex and flexible harmonic spectra. The principles behind oscillator synchronization are described in Synchronizing the ES2 Oscillators. Ring modulation principles are discussed in Using Ring Modulation in the ES2.

Using Pulse Width Modulation in the ES2

You can alter the tonal color of rectangular waveforms by scaling the width of waveform pulses to any value. This is known as pulse width modulation.

The pulse width modulation features of the ES2 are extensive. As an example, if rectangular waves are chosen for all oscillators, you can simultaneously modulate the pulse width of Oscillator 1 and the synchronized pulse waves of Oscillator 2 (or the square wave of Oscillator 2's ring modulator) and Oscillator 3.

To set a basic pulse width in Oscillator 2 or 3

- Drag the waveform rotary control in the area highlighted in the image above.

  Only Oscillators 2 and 3 allow you to define a "base" pulse width, prior to any pulse width modulation.

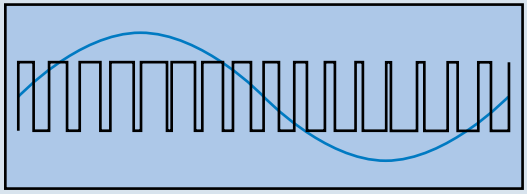
To set up a pulse width modulation routing (of Oscillator 1) in the router:

1. Choose Osc1Wave as the Target.
2. Choose LFO1 as the Source.
3. Adjust the modulation amount slider (a value of 0.12 is subtle, but sweet).
4. Choose a sine wave for LFO 1.
5. Adjust the LFO 1 Rate (0.160 Hz is nice for slow sweeps).

Modulating ES2 Oscillator Pulse Width with an LFO

Pulse width modulation (PWM) can be automatically controlled with appropriate settings in the router. A pulse wave (with PWM controlled by an LFO set to a sine wave) makes a single oscillator sound vivid, undulating, and overtone-rich. Sonically, this is similar to the sound of two slightly detuned, phasing oscillators, which is great for sustained bass and pad sounds.

The graphic below shows a pulse wave, with the pulse width modulated by an LFO. You can clearly see how the width of the pulses changes over time.



Note: Select the intensity and speed of the modulation with care, because the overall volume decreases, and slight detuning occurs, when the pulses become very narrow (below a width value of 10%).

Tip: Pulse width modulations via velocity-sensitive envelope generators sound very dynamic—a great effect that is especially suitable for percussive bass sounds.

Using Frequency Modulation in the ES2

The principle of frequency modulation (FM) synthesis was developed in the late 1960s and early 1970s by John Chowning. It was popularized by Yamaha's range of DX synthesizers in the 1980s. The ES2 can't be compared with the DX series in the discipline of pure FM synthesis, but it can certainly achieve some of the signature sounds of these instruments.

How Frequency Modulation Works

Stated simply, the frequency of one signal generator, or oscillator, is altered—modulated—by another signal generator. Positive values from the second generator increase the frequency of the first generator. Negative values decrease the frequency.

In a synthesizer, this type of modulation takes place in the audible range. Depending on the design of the instrument, you can hear the signals of either the first oscillator alone (being modulated by the other oscillator), or both oscillators. The interaction between the two generators alters the waveform signal of the first oscillator and introduces a number of new harmonics. This harmonic spectrum can then be used as the source signal for further sound processing, such as filtering, envelope control, and so on. See Frequency Modulation (FM) Synthesis for further information.

How Frequency Modulation Works in the ES2

In the ES2, the frequency of Oscillator 1 (with a sine wave chosen—11 o'clock position for the Wave knob) can be modulated by the output signal of Oscillator 2.

- When Oscillator 2 outputs a positive signal, the frequency of Oscillator 1 increases.
- When Oscillator 2 outputs a negative signal, the frequency of Oscillator 1 decreases.

The net effect of speeding up or slowing down the frequency of Oscillator 1 in each waveform cycle is a distortion of the basic wave shape. This waveform distortion also has the side benefit of introducing a number of new, audible harmonics.

Important: The impact of any frequency modulations you perform depends on *both* the frequency ratio *and* the modulation intensity of the two oscillators.

To set the frequency ratio and adjust the modulation intensity

1. Adjust the Frequency (coarse and fine tune) parameter values of one, or both, oscillators.
2. Click (or drag) in the control range between the Sine and FM icons around the Oscillator 1 Wave knob. This determines the amount, or intensity, of frequency modulation.

FM Synthesis Options in the ES2 Router

You can also adjust the FM intensity by choosing the Osc1Wave modulation target in the router.

- Use lower FM intensity values if you are after a subtle FM effect.
- If you're after more extreme FM modulations, choose the Osc1WaveB target in the router. See ES2 Modulation Target Reference for further details.

Using Different ES2 Waveforms for FM Synthesis

The "pure" FM synthesis method uses a sine wave for both the first and second signal generator (both Oscillator 1 and 2 would be limited to generating a sine wave in the ES2 if you stuck with this approach).

The ES2, however, provides 100 Digiwaves and countless combinations of modulation intensities and frequency ratios that can be used for either oscillator. This provides a vast pool of harmonic spectra and tonal colors for you to experiment with—so it would be a shame if you don't take advantage of these options!

Tip: The type of modulation that occurs can vary significantly when different waveforms are chosen for Oscillator 2—the modulating oscillator—in particular.

Using Ring Modulation in the ES2

Ring modulation is a powerful tool for the creation of inharmonic, metallic, bell-like sounds. The spectra resulting from its use are inharmonic at almost every frequency ratio. The ring modulator is a device that dates back to the early days of the synthesizer.

How a Ring Modulator Works

A ring modulator has two inputs. At the output you hear both the sum and difference frequencies of the input signals. If you ring modulate a sine oscillation of 200 Hz with a sine oscillation of 500 Hz, the output signal of the ring modulator will consist of a 700 Hz (sum) and a 300 Hz (difference) signal. Negative frequencies result in a change to the phase polarity of output signals.

How Ring Modulation Works in the ES2

You can set Oscillator 2 to output a ring modulator signal by choosing the Ring setting with the Oscillator 2 Wave knob. Feel free to experiment with different Frequency (main and fine tune) values for one, or both, oscillators.

The Oscillator 2 ring modulator is fed with the output signal of Oscillator 1 and a square wave—generated by Oscillator 2 itself. The pulse width of this square wave can be modulated (see Using Pulse Width Modulation in the ES2).

Tip: Use sawtooth and rectangular (pulse width modulated) input signals from Oscillators 1 and 2, respectively, to create a much more complex output signal. The use of these harmonically rich waveforms results in a number of extra sidebands becoming audible.

Using Digiwaves in the ES2

In addition to the basic synthesizer waveforms, all of the ES2 oscillators provide 100 additional waveforms, called Digiwaves. These are very short samples of the attack transients of various sounds and instruments.

To choose a Digiwave

- Set the Wave knob to Sine (6 o'clock position), then do one of the following:
- Control-click or right-click the Sine label, and choose a waveform from the pop-up menu.
- Click-hold the Sine label and drag the mouse vertically.
- Shift-click the menu, and type in a value, to select the Digiwave numerically.

ES2 Digiwave Modulation Options

The number assigned to each Digiwave is a parameter that can be modulated. This feature allows the ES2 to produce sounds reminiscent of classic wavetable synthesizers from PPG and Waldorf. See Wavetable, Vector, and Linear Arithmetic (LA) Synthesis.

You can automatically scroll through the list of Digiwaves by modulating the OscWave target in the modulation router. If you choose a sufficiently low modulation intensity and speed, you will hear each individual Digiwave being cross-faded.

The Digiwaves of the three oscillators can be modulated independently or as a group. These modulation targets are discussed in Getting to Know the ES2 Oscillators.

Using Noise in the ES2 (Oscillator 3 Only)

The sonic palette of Oscillator 3 is bolstered by the inclusion of a noise generator, which can be activated by choosing the noise waveform. By default, Oscillator 3's noise generator generates *white noise*.

White noise is defined as a signal that consists of all frequencies (an infinite number) sounding simultaneously, at the same intensity, in a given frequency band. The width of the frequency band is measured in Hertz. Sonically, white noise falls between the sound of the consonant "*F*" and breaking waves (surf). White noise is useful for synthesizing wind and seashore noises, or electronic snare drum sounds.

Modulating the Noise Color

Oscillator 3 has more up its sleeve than the output of neutral sounding white noise, however. You can modulate the tonal color of the noise signal in real time—without using the main filters of the ES2—by modulating the waveform of Oscillator 3.

To change the noise color, set up a modulation routing as follows: modulation target Osc3Wave, source ModWhl. The modulation amount slider behaves somewhat differently with this routing, essentially acting like a filter.

- Use negative modulation amount values (not –1.000) to set a descending filter slope that roughly equates to 6 dB/octave, and the sound becomes darker (red noise) as you adjust the mod wheel downwards.
- You can effectively tune this pseudofilter down to 18 Hz, with a modulation amount setting of –1.000. When Osc3Wave is modulated positively, the noise becomes brighter (blue noise).
- If you choose a modulation amount value of +1.000 for the Osc3Wave modulation target, the filter cutoff frequency is set to 18 kHz.

Emulating Detuned Analog Synthesizer Oscillators in the ES2

The Analog parameter is found towards the top left of the ES2 interface. It alters the pitch of each note—and the filter cutoff frequency—in a random fashion.

Much like polyphonic analog synthesizers, all three oscillators maintain their specific frequency deviation from each other, but the pitches of all three oscillators are randomly detuned by the same Analog amount. For example, if the Analog detuning is set to around 20%, all three oscillators (if used) will randomly drift by 20%.

- Low Analog values can add a subtle richness to the sound.
- Medium Analog values simulate the tuning instabilities of analog synthesizer circuitry, which can be useful in achieving that much sought-after "warmth" of analog hardware synthesizers.
- High Analog values result in significant pitch instability, which can sound truly out of tune—but this may be perfect for your needs.

Note: If the ES2 is set to Mono or Legato keyboard mode, the Analog parameter is effective only when Unison is active. In this situation, Analog sets the amount of detuning between the stacked (unison) voices. If the Voices parameter is set to 1 and/or Unison is not activated, the Analog parameter has no effect. For more information about these parameters, see Choosing the ES2 Keyboard Mode (Poly/Mono/Legato).

Emulating Stretch Tuning in the ES2

The (coarse) Frequency knob of each oscillator allows you to tune Oscillators 1, 2, and 3 in semitones or octaves. The (fine tune) Frequency parameter allows you to fine-tune each oscillator in cents (1/100th of a semitone). Precise detuning between oscillators can result in beats, or phasing, between the oscillator frequencies. The higher the played frequency/pitch, the faster the phasing beats. High notes, therefore, may seem to be somewhat out of tune in comparison with lower notes.

The CBD (Constant Beat Detuning) parameter—to the left of Oscillator 2's Frequency knob—detunes the harmonics of the low-note frequencies in a ratio that is proportionate with the fundamental tone of the upper-note frequencies.

This is a very natural-sounding effect, common to acoustic pianos, which are intentionally tuned "out-of-tune" (from equal temperament). This is known as *stretch tuning*—and results in the upper and lower keyboard ranges being slightly out of tune with the center octaves but harmonically "in-tune" with each other.

Some Tips on Using Constant Beat Detuning

CBD can be used as a corrective tool to even out the beating between oscillators, or it can be used as a creative tool to emulate stretch tuning. The latter can be particularly important when you use an ES2 sound alongside an acoustic piano recording.

The CBD parameter offers five values: off, 25%, 50%, 75%, and 100%. If you choose 100%, the phasing beats are almost constant across the entire keyboard range. This value may, however, be too high, as the lower notes might be overly detuned at the point where the phasing of the higher notes feels right. Try lower CBD values in cases where the bass notes appear to be a little too far out of tune with the upper keyboard range.

The reference pitch for CBD is C3 (middle C): Its (de)tuning is constant, regardless of the chosen CBD value.

Setting the Oscillator Level Balance in the ES2

Dragging the square icon in the Triangle cross-fades—sets the level relationships—between the three oscillators. This is self-evident in use. If you move the square icon along one of the Triangle's sides, it cross-fades between the two closest oscillators, and the third oscillator is muted.

Click or click-drag in the Triangle to change the level balance between the oscillators.

Modulating Triangle Coordinates with the ES2 Router

The position of the square icon in the Triangle is described by two parameters—coordinates, actually—which are relevant when it comes to automating the oscillator mix. These parameters, called OscLevelX and OscLevelY, are accessible as targets in the router.

Important: Don't confuse these coordinates with the X and Y positions of the Planar Pad (see Using the ES2's Planar Pad).

Controlling Triangle Coordinates with the ES2 Vector Envelope

The position of the square icon in the Triangle can be controlled via the Vector Envelope. As the Vector Envelope features a loop function, it can be used as a pseudo-LFO with a programmable waveform. This can be used to alter the positions of the square icon in the Triangle. For more information about this feature, see Vector Envelope Control of the ES2's Planar Pad and Triangle, and Getting to Know the ES2's Vector Envelope.

Adjusting the ES2 Oscillator Start Point

The oscillators can either run freely or begin at the same phase position of their respective waveform cycles—each time the ES2 receives a note on message. You can set the desired behavior using the Osc Start (Oscillator Start) pop-up menu, found at the upper-right corner of the ES2 interface.

- When Osc Start is set to free: The initial oscillator phase start point is random for each played note. This adds life to the sound. The downside is that the output level may differ each time a note is played, making the attack phase sound less punchy—even if the performance is identical each time—such as when the note is triggered by a MIDI region. This setting is useful when you are emulating the sounds typical of hardware analog synthesizers.
- When Osc Start is set to soft: The initial oscillator phase starts at a zero crossing for each played note. This mimics the sonic character (and precision) that is typical of digital synthesizers.
- When Osc Start is set to hard: The initial oscillator phase starts at the highest level in the waveform cycle for each played note. The extra "punch" that this setting can provide is audible only if the ENV3 Attack Time parameter is set to a low value—a very fast attack, in other words. This setting is highly recommended for electronic percussion and hard basses.

Note: Osc Start soft and hard result in a constant output level of the initial oscillator phase every time the sound is played back. This may be of particular importance when you use the Bounce function of Logic Express at close to maximum recording levels.

Synchronizing the ES2 Oscillators

Typical "oscillator sync" sounds tend toward the aggressive, screaming leads that synthesizer manufacturers like to talk about. The rectangular and sawtooth waveforms of Oscillators 2 and 3 feature a Sync option. When this parameter is turned on, the phase of Oscillator 2 or 3 is synchronized with Oscillator 1.

Every time Oscillator 1 starts a new oscillation phase, the synchronized oscillator (Oscillator 2 or 3) is also forced to restart its phase from the beginning. Between the waveform cycles of Oscillator 1, the waveform cycles of the synchronized oscillators run freely.

Envelope Modulation of the Synchronized Oscillator Frequency

Synchronized oscillator sounds are especially cool when the frequency of the synchronized oscillator is modulated by an envelope generator. This way, the number of phases within a section of the synchronization cycle constantly changes, resulting in corresponding changes to the frequency spectrum.

Using the ES2's Global Parameters

These parameters affect the overall instrument sound produced by the ES2. You can find the global parameters to the left of the oscillators, and above the filter and output sections.

- *Keyboard Mode buttons:* These buttons switch the ES2 between polyphonic, monophonic, and legato behaviors. See Choosing the ES2 Keyboard Mode (Poly/Mono/Legato).
- *Unison button:* This button enables or disables unison mode. See Using Unison and Voices for a Richer ES2 Sound.
- *Voices field:* Determines the maximum number of notes that can be played simultaneously.
- *Glide knob:* This parameter—also known as *Portamento*—determines the amount of time it takes for the pitch of a played note to slide to the pitch of the following played note. See Setting a Glide (Portamento) Time in the ES2.
- *Bend Range fields:* This parameter pair determines the upward and downward pitch bend range. See Determining a Bend Range in the ES2.
- *Tune field:* Sets the pitch of the ES2 in cents. 100 cents equals a semitone step. At a value of 0 c (zero cents), the central A key is tuned to 440 Hz, or concert pitch.
- *Analog knob:* See Emulating Detuned Analog Synthesizer Oscillators in the ES2).

- *Constant Beat Detuning (CBD) menu:* See Emulating Stretch Tuning in the ES2.
- *Oscillator Start menu:* See Adjusting the ES2 Oscillator Start Point.

Choosing the ES2 Keyboard Mode (Poly/Mono/Legato)

A *polyphonic* instrument allows several notes to be played simultaneously—an organ or piano, for example. Many older analog synthesizers are *monophonic*, which means that only one note can be played at a time, much like a brass or reed instrument. This shouldn't be viewed as a disadvantage in any way, because it allows playing styles that are not possible with polyphonic instruments.

- If you choose Mono mode, staccato playing will retrigger the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato, and then they continue their curve until you release the last legato played key.
- The Legato mode is also monophonic, but with one difference: The envelope generators are retriggered only if you play staccato—releasing each key before playing a new key.

Note: On several monophonic synthesizers, the behavior in Legato mode is referred to as *single trigger*, while Mono mode is referred to as *multi trigger*.

Using Unison and Voices for a Richer ES2 Sound

One of the great strengths of polyphonic analog synthesizers is unison—or "stacked voices"—mode. Traditionally, in unison mode classic analog polysynths run monophonically, with all voices playing simultaneously when a single note is struck. As the voices of an analog synthesizer are never perfectly in tune, this results in an extremely fat chorus effect with great sonic depth.

To enable monophonic unison mode for the ES2

- Activate either Mono or Legato mode, and turn on the Unison button:
  - The intensity of the unison effect depends on the number chosen in the Voices parameter field. Increase the Voices value for a fatter sound.
  - The intensity of detuning (voice deviation) is set with the Analog parameter (see Emulating Detuned Analog Synthesizer Oscillators in the ES2).

To use the ES2 in polyphonic unison mode

- Activate the Poly and Unison buttons.

In Poly/Unison mode, each played note is effectively doubled—or, more correctly, the polyphony value of the Voices parameter is halved. These two voices are then heard when you trigger the note. Activating Poly/Unison has the same effect as setting the ES2 to Mono/Unison (Voices = 2), but you can play polyphonically.

Setting a Glide (Portamento) Time in the ES2

The Glide parameter controls the portamento time. This is the amount of time it takes for the pitch of one played note to travel to the pitch of another played note.

The Glide parameter behavior depends on the keyboard mode you choose. See Choosing the ES2 Keyboard Mode (Poly/Mono/Legato).

- If the keyboard mode is set to Poly or Mono, and Glide is set to a value other than 0, portamento is active.
- If Legato is chosen, and Glide is set to a value other than 0, you need to play legato (pressing a new key while holding the old one) to activate portamento. If you don't play in a legato style, portamento won't work. This behavior is also known as *fingered portamento*.

Determining a Bend Range in the ES2

The Bend range fields determine the range for pitch bend modulation, typically performed with your keyboard's pitch bend wheel.

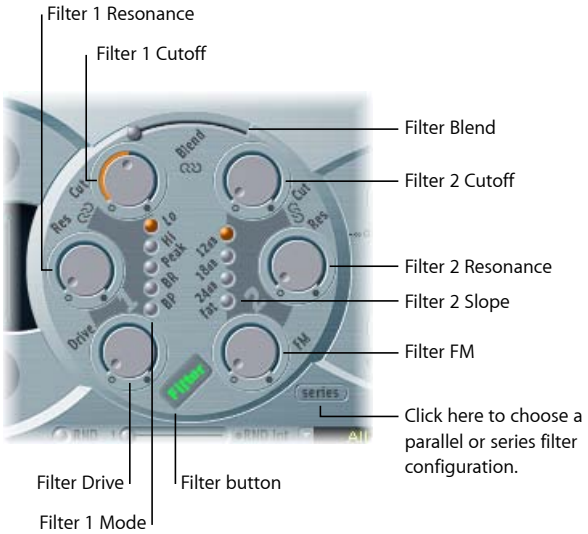
There are separate range fields for upward and downward bends. When you set the right-hand Bend range field to Link mode, the bend range is identical in both directions—if you assign a downward bend of 4 semitones, this will be mirrored in the upward bend field, resulting in a combined bend range of 8 semitones—9 if you include the standard pitch, or "no bend" position.

Using the ES2 Filters

The ES2 features two discrete, and different, filters.

- Filter 1 can operate as a lowpass, highpass, bandpass, band reject, or peak filter.
- Filter 2 is a lowpass filter that offers variable slopes (measured in dB/octave).

Details on all filter parameters are covered in the following sections.



- *Filter button:* Activates or deactivates the entire filter section of the ES2. Deactivating the filter section makes it easier to hear adjustments to other sound parameters, because the filters always heavily affect the sound. Disabling the filters also reduces processor load. If the Filter label is green, the filters are engaged. If gray, the filters are disabled.
- *Filter Configuration button:* Switches between a parallel or series filter configuration. See Choosing a Series or Parallel Filter Configuration in the ES2.
- *Filter Blend slider:* Sets the balance between Filter 1 and Filter 2. See Filter Blend: Cross-Fading Between the ES2 Filters.
- *Filter 1 Mode buttons:* These buttons switch Filter 1 between lowpass, highpass, bandpass, band reject, or peak filter types. See Choosing Filter 1's Mode in the ES2 (Lo, Hi, Peak, BR, BP).
- *Filter 2 Slope buttons:* These buttons switch Filter 2 between different slopes. See Setting Filter 2's Slope in the ES2.
- *Cutoff and Resonance:* The Cutoff and Resonance knobs determine the cutoff frequency and resonance behavior of each filter. See Using the ES2 Filter Cutoff and Resonance Parameters.
- *Filter Drive knob:* Overdrives the filter, which affects each voice independently. See Overdriving the ES2 Filters.
- *Filter FM knob:* Modulates the Filter 2 Cutoff parameter with the Oscillator 1 frequency. See Modulating Filter 2's Frequency in the ES2.

Choosing a Series or Parallel Filter Configuration in the ES2

Click the Parallel/Series button to switch between a parallel and series filter routing. When either is chosen, the entire circular filter element of the ES2 user interface rotates, and the positions and direction of the filter controls clearly indicate the signal flow. The button name also changes in each mode.

Series filter signal flow

Parallel filter signal flow

In the figure to the left, the filters are cabled in series. This means that the signal of all oscillators (combined at the Oscillator Mix Triangle) passes through the first filter, and then this filtered signal passes through Filter 2—if Filter Blend (see below) is set to 0, the middle position. The output signal of Filter 2 is then sent to the input of the dynamic stage (Amplifier section).

In the figure to the right, the filters are cabled in parallel. If Filter Blend is set to 0, you'll hear a 50/50 mix of the source signal—routed via Filter 1 and Filter 2. The output signals of the two filters are then sent to the input of the dynamic stage.

Filter Blend: Cross-Fading Between the ES2 Filters

You can use the Filter Blend slider to cross-fade between the two filters when cabled in parallel. Filter Blend can have a significant effect on the ES2 signal flow. See The Impact of Filter Blend on the ES2 Signal Flow.

- When Filter Blend is set to its top position, you'll only hear Filter 1.
- If Filter Blend is set to its lowest position, you'll only hear Filter 2.

- In between these positions, the filters are cross-faded.

You can also cross-fade the filters when they are cabled in series. In this scenario, the distortion—controlled by the Drive parameter—also needs to be considered, as this can be positioned either before or in-between the filters, depending on the Filter Blend setting you choose.

Using the ES2 Router to Control Filter Blend

The Filter Blend parameter is available as a modulation target in the router. You can obviously use manual control sources, such as the modulation wheel, to change the filter blend; but the Filter Blend can also be used creatively, to rapidly switch or smoothly fade between the two filters.

Examples of sources you might consider include an LFO, velocity, or even a combination of the Vector Envelope and Planar Pad—using the X or Y Pad parameters as sources. The latter would enable some interesting filter control possibilities that would evolve along with oscillator parameters (also being controlled via the Vector Envelope), or independently.

The Impact of Filter Blend on the ES2 Signal Flow

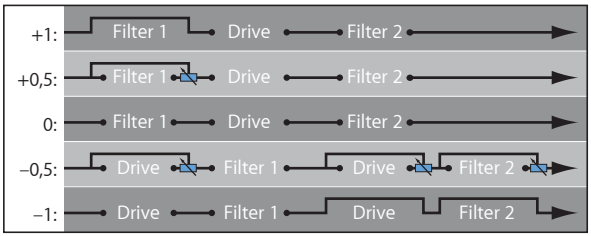
Regardless of whether parallel or series filter configurations are chosen, a Filter Blend setting of –1 results in only Filter 1 being audible. A Filter Blend setting of +1 will limit audibility to Filter 2.

The figures illustrate the signal flow between the Oscillator Mix stage (the Triangle) and the dynamic stage. The signal flow through the filters and the filter overdrive circuit (the Drive parameter) are dependent on the Filter Blend setting.

Filter Blend: Series Filter Configuration Information

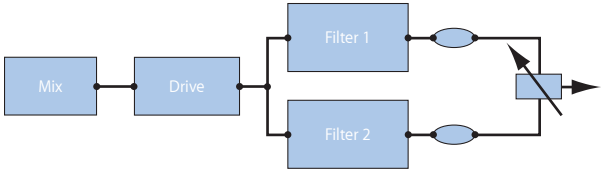
- Use *positive* values for Filter Blend to partially bypass Filter 1.
- Use *negative* values for Filter Blend to partially bypass Filter 2.
- When *zero* or *positive* Filter Blend values are used, there is only one overdrive circuit for *both* filters.
- Use of *negative* Filter Blend values introduces another overdrive circuit, which distorts the output signal of the oscillator mix stage before it is fed into the first filter.

- If Drive is set to 0, no distortion occurs.



Filter Blend: Parallel Filter Configuration Information

In a parallel configuration, the overdrive/distortion circuit—the Drive parameter—is always wired *after* the oscillator mix stage—the Triangle—and *before* the filters. The filters receive a mono input signal from the output of the overdrive circuit. The outputs of both filters are mixed to mono via Filter Blend.



Choosing Filter 1's Mode in the ES2 (Lo, Hi, Peak, BR, BP)

Filter 1 can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.

Select one of the following buttons to choose a filter mode for Filter 1:

- *Lo (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. When set to Lo, the filter operates as a lowpass filter. The slope of Filter 1 is fixed at 12 dB/octave in Lo mode.
- *Hi (highpass):* This filter type allows frequencies above the cutoff frequency to pass. When set to Hi, the filter operates as a highpass filter. The slope of Filter 1 is fixed at 12 dB/octave in Hi mode.
- *Peak:* Filter 1 works as a *peak filter*. This allows the level in a frequency band to be increased. The center of the frequency band is determined by the Cutoff parameter. The width of the band is controlled by the Resonance parameter.

- *BR: (band rejection):*The frequency band directly surrounding the cutoff frequency is rejected, whereas the frequencies outside this band can pass. The Resonance parameter controls the width of the rejected frequency band.
- *BP (bandpass):* The frequency band directly surrounding the cutoff frequency is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the center frequency of the band.

Setting Filter 2's Slope in the ES2

Most filters do not completely suppress the portion of the signal that falls outside the frequency range defined by the Cutoff parameter. The slope, or curve, chosen for Filter 2 expresses the amount of rejection below the cutoff frequency in decibels per octave.

Filter 2 offers three different slopes: 12 dB, 18 dB, and 24 dB per octave. The steeper the slope, the more severely the level of signals below the *cutoff frequency* are affected in each octave.

The Fat setting also provides 24 dB per octave of rejection but features a built-in compensation circuit that retains the "bottom end" of the sound. The standard 24 dB setting tends to make lower end sounds somewhat "thin." See Getting to Know the ES2 Oscillators.

Using the ES2 Filter Cutoff and Resonance Parameters

In every lowpass filter (in the ES2:Lo mode for Filter 1. Filter 2 is a lowpass filter), all frequency portions above the cutoff frequency are suppressed, or cut off, hence the name. If you're new to synthesizers and the concepts behind filters, see Synthesizer Basics.

The Impact of Cutoff Frequency on the ES2 Signal

The Cutoff Frequency (Cut) parameter controls the brilliance of the signal.

- In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
- In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed and only upper frequencies are allowed to pass.
- In a bandpass/band rejection filter, the cutoff frequency determines the center frequency for the bandpass or band rejection filter.

The Impact of Resonance on the ES2 Signal

The Resonance (Res) parameter emphasizes or suppresses portions of the signal above or below the defined cutoff frequency.

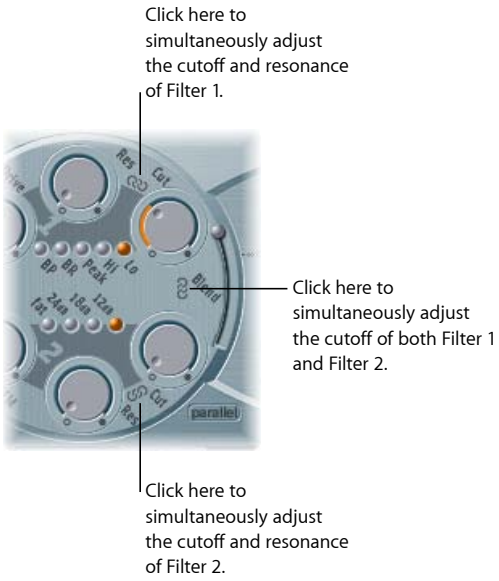
- In a lowpass filter, Resonance emphasizes or suppresses signals below the cutoff frequency.
- In a highpass filter, Resonance emphasizes or suppresses signals above the cutoff frequency.
- In bandpass/band rejection filters, resonance emphasizes or suppresses the portions of the signal—the frequency band—that surround the defined frequency, set with the Cutoff Frequency parameter.

Controlling Cutoff and Resonance Simultaneously in the ES2

Changing the Cutoff and Resonance controls at the same time is essential to the creation of expressive synthesizer sounds.

To control two filter parameters at once

- Drag one of the three chain symbols in the ES2 filter section.



- The chain between Cut and Res of Filter 1 controls both the resonance (drag horizontally) and cutoff frequency (drag vertically) simultaneously.
- The chain between Cut and Res of Filter 2 controls both the resonance (drag horizontally) and cutoff frequency (drag vertically) simultaneously.
- The chain between Filter 1 Cut and Filter 2 Cut controls the cutoff frequency of Filter 1 (drag vertically) and Filter 2 (drag horizontally) simultaneously.

Using Flt Reset to Drive the ES2 Filters to Self Resonance

If you increase the filter Resonance parameter to higher values, the filter will begin to internally feed back and, as a consequence, will begin to self-resonate. This self-resonance results in a sine oscillation—a sine wave—that is actually audible.

To start this type of oscillation, the filter requires a trigger. In an analog synthesizer, this trigger can be the noise floor, or the oscillator output. In the digital domain of the ES2, noise is all but eliminated. Therefore, when the oscillators are muted there is no input signal routed to the filter.

To drive the ES2 filter to self-resonance

- Turn on the Filter Reset button in the upper-right corner of the ES2 interface.

When engaged, each note starts with a trigger that makes the filter resonate immediately.

Using the ES2's Fat Parameter to Compensate for High Resonance Values

Increasing the Resonance value results in a rejection of bass—low frequency energy—when you are using lowpass filters.

Turn on the Fat(ness) button—below the other filter slope buttons—to compensate for this side-effect and to obtain a richer sound.

Overdriving the ES2 Filters

The filters are equipped with separate overdrive modules. Overdrive intensity is defined by the Drive parameter.

- If the filters are connected in parallel, the overdrive circuit is placed before the filters.
- If the filters are connected in series, the position of the overdrive circuits is dependent on the Filter Blend parameter—as described in Filter Blend: Cross-Fading Between the ES2 Filters.

The ES2 filter Drive parameter affects each voice independently. When every voice is overdriven individually—like having six fuzz boxes for a guitar, one for each string—you can play extremely complex harmonies over the entire keyboard range. They'll sound clean, without unwanted intermodulation effects spoiling the sound.

Furthermore, appropriate Drive parameter settings lead to a different tonal character for the following reason: The way analog filters behave when they are overdriven forms an essential part of the sonic character of a synthesizer. Each synthesizer model is unique in the way its filters behave when overdriven. The ES2 is very flexible in this area, allowing tonal colors that range from the most subtle fuzz to the hardest of distortions.

Tip: As Filter 2 can cut away the overtones introduced by the distortion, the Drive parameter can be seen and used as another tool for deforming oscillator waveforms.

Polyphonic Distortions in the Real World

The ES2 features a dedicated *distortion* effect in the Effects section. Given the inclusion of this effect, you may be wondering what benefit the Drive function in the Filter section brings.

The Distortion circuit in the Effects section affects the entire polyphonic output of the ES2. Every rock guitarist knows that more complex chords—other than major chords, parallel fifths, and octaves—sound "rough," when using distortion. Therefore, distorted guitar playing generally involves few voices, or parallel fifths and octaves.

The ES2 filter Drive parameter affects each voice individually, allowing you to play complex chords without introducing the unpleasant intermodulations that the Distortion effect can add to your sound.

Modulating Filter 2's Frequency in the ES2

The Filter 2 cutoff frequency can be modulated by the sine wave of Oscillator 1, which is always generated, even when the oscillator is switched off. The level of this sine signal can be mixed in at the output stage with the Sine Level parameter (see Thickening the ES2's Sound with Sine Level).

The effect of such filter modulations in the audio spectrum is unpredictable, but the results tend to remain harmonic if you avoid high modulation intensity values. The FM parameter is used to define the intensity of this filter frequency modulation.

Note: Don't confuse this filter frequency modulation with the oscillator FM feature (Oscillator 1 is modulated by Oscillator 2, as described in Using Frequency Modulation in the ES2). If Oscillator 1 is frequency-modulated by Oscillator 2, it does not influence the sine wave signal used to modulate the cutoff frequencies.

Controlling Filter FM in the ES2 Router

You can alter the value of the filter FM parameter with an appropriate modulation routing in the router.

Choose LPF FM as the modulation target. A sine wave, at the frequency of Oscillator 1, is always used as the modulation source.

Given the default assignment of Oscillator 1 as the modulation source, and the direct relationship between the filter FM intensity and Oscillator 1's frequency, you can set up a second routing with Oscillator 1 frequency (Pitch1) as the target.

Filter 2 can also be driven to self-oscillation. If you set a very high Resonance value, it will produce a sine wave. This self-oscillating sine wave will distort at the maximum Resonance value. If you mute all oscillators, you'll only hear this sine oscillation. By modulating the Cutoff Frequency, you can produce effects similar to those produced by modulating the frequency of Oscillator 1 with Oscillator 2.

Using the ES2 Amplifier Parameters

The dynamic stage of a synthesizer defines the level—the perceived volume—of a played note. The change in level over time is controlled by an *envelope generator*. For more information about envelope generators, see Synthesizer Basics.

Using Envelope 3 to Control ES2 Levels

ENV 3 is hard-wired to the dynamic stage of the ES2—it is always used to control the level of each note. All envelope parameters are discussed in Getting to Know the ES2 Envelopes (ENV 1 to ENV 3), with specific information about Envelope 3 found in Getting to Know ENV 2 and ENV 3 in the ES2.

Modulating the Amp in the ES2 Router

The dynamic stage can be modulated by any router modulation source. To do this, choose the AMP modulation target in a modulation routing in the router.

For example, you can quickly create a tremolo effect by selecting AMP as the target, and LFO1 as the source (leave via set to Off) in a modulation routing. The level will change periodically, based on the current LFO 1 Rate value.

Thickening the ES2's Sound with Sine Level

The Sine Level knob, located next to the Filter 2 section, allows a sine wave at the frequency of Oscillator 1 to be mixed directly into the dynamic stage, independent of the filters. Even if you have filtered away the basic partial tone of Oscillator 1 with a highpass filter, you can reconstitute it with this parameter.

- In cases where Oscillator 1 is frequency-modulated by Oscillator 2 (if you have turned up FM with the waveform selector), *only* the pure sine wave is mixed into the dynamic section, not the distorted FM waveform.
- Any modulation of Oscillator 1's pitch, set in the router, affects the frequency of the sine wave mixed in at this stage.

Note: The Sine Level knob is useful for adding warmth and a fat bass quality to the sound. Extra body can be added to thin sounds with this function, given that Oscillator 1 actually plays the basic pitch.

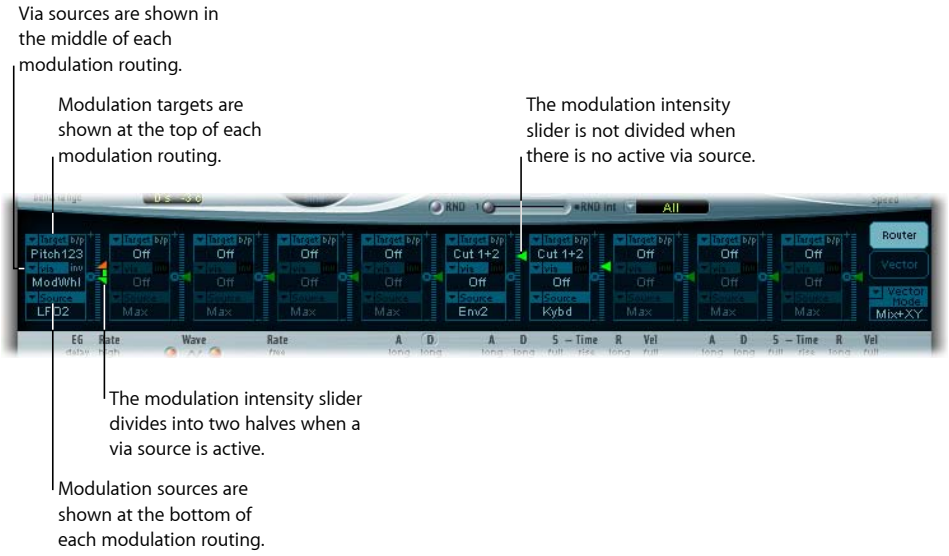
Working with Modulation in the ES2

The ES2 is equipped with a huge number of modulation sources and targets, making the ES2 a very flexible synthesizer that can generate extraordinary sounds that constantly evolve, sound like audio loops, or are just plain expressive to play. Reference tables that cover all modulation targets and sources are found at the end of this section.

- *Modulation router:* The modulation router—or router, for short—links modulation sources such as the envelope, to modulation targets such as the oscillators and filters. The router features 10 modulation routings, arranged into columns. See Getting to Know the ES2's Modulation Router.
- *Modulation sources:* The modulation sources include the LFOs and envelopes. See Getting to Know the ES2 LFOs and Getting to Know the ES2 Envelopes (ENV 1 to ENV 3).
- *Vector Envelope:* The Vector Envelope is an extremely sophisticated, loop-capable, multipoint envelope that can control the Planar Pad and Triangle (oscillator mix parameter). The Vector Envelope shares the space occupied by the modulation router and can be viewed by clicking the Vector Envelope button to the right of the router. See Getting to Know the ES2's Vector Envelope.
- *Planar Pad:* The Planar Pad is a two-dimensional controller—which facilitates the simultaneous manipulation of two, freely assignable, parameters. It can be controlled with the Vector Envelope. See Using the ES2's Planar Pad.

Getting to Know the ES2's Modulation Router

The modulation router—or router—spans the center of the ES2 interface. Click the Router button to view it if the Vector Envelope is displayed (these components share the same section of the interface). If you are new to synthesizer modulation routings, refer to the Synth Basics Modulation section.


Via sources are shown in the middle of each modulation routing.
Modulation targets are shown at the top of each modulation routing.
The modulation intensity slider is not divided when there is no active via source.
Pitch 123
ModWhl
LFO2
Cut 1+2
Cut 1+2
Env2
Kybd
Router
Vector
The modulation intensity slider divides into two halves when a via source is active.
Modulation sources are shown at the bottom of each modulation routing.

Any modulation *source* can be connected to any modulation *target*—much like an old-fashioned telephone exchange or a studio patch bay.

The modulation intensity—how strongly the target is influenced by the source—is set with the vertical slider to the right of the modulation routing.

The intensity of the modulation can itself be modulated: The *via* parameter defines a further modulation source, which is used to control the modulation intensity. When via is active, you can specify upper and lower limits for the modulation intensity.

Ten such modulation routings of source, via, and target can take place simultaneously, in addition to routings that are hard-wired outside of the router. It does not matter which of the ten modulation routings you use.

You can even select the same target in several parallel modulation routings. You can also use the same sources and the same via controllers in multiple modulation routings.

Creating and Bypassing ES2 Modulation Routings

The following information applies to all ten modulation routings.

To create a basic modulation routing

1 Click in the Target field. A pop-up menu of all available targets appears.

2 Click in the Source field. A pop-up menu of all available sources appears.

3 Choose the parameter you want to modulate.

Choose the parameter you want to use to modulate the target.

4 Vertically drag the arrowhead of the Intensity slider to the right of the modulation routing.

This sets a fixed modulation intensity.

To bypass a modulation routing

- Click the "b/p" button at the top right of the modulation routing beside the Target label.

The Bypass (b/p) parameter allows you to enable or disable individual modulation routings, without losing settings.

Using Via Sources to Control ES2 Modulation Intensity

In a basic modulation routing comprised of a target and source, you can set a fixed modulation intensity by vertically dragging the arrowhead of the Intensity slider to the right of the routing. The slider value always defines a constant modulation intensity.

The intensity of the modulation can itself be modulated: The *via* parameter defines a further modulation source, which is used to control the modulation intensity. As soon as you choose a value other than off for "via," the Intensity slider is divided into two halves. Each half has its own arrowhead.

- The lower half defines the minimum modulation intensity, when the via controller—the modulation wheel, for example—is set to its minimum value.
- The upper half defines the maximum modulation intensity when the via controller is set to its maximum value.
- The area between the two slider halves defines the modulation range of the via controller.

To create a modulation routing that includes a via source

1. Click in the Target field. A pop-up menu of all available targets appears.

2. Click in the Source field. A pop-up menu of all available sources appears.

3 Click-hold in the via field to see a pop-up menu of all available sources.

4 Choose the source you want to use for control of the modulation intensity.

5 Vertically drag the upper arrowhead of the Intensity slider (to the right of the modulation routing) to set the maximum modulation intensity.

6 Vertically drag the lower arrowhead of the Intensity slider to set the minimum modulation intensity.

To move the entire via range

- Drag the range area between the two slider halves vertically.

Both arrowheads will move simultaneously.

If this area is too small to be dragged, just drag an unused section of the Intensity slider "track" to move the area.

To set the modulation intensity to zero

- Click the little zero symbol beside the via label.

To invert the effect of the via modulation source

- Click the via invert (inv) parameter to the right of the via label.

An ES2 Modulation Example

Imagine an ES2 sound with these settings:

- *Target:* Pitch123
- *via:* ModWheel
- *Source:* LFO1
- *Modulation intensity:* Slider position, set as desired

Pitch123 is the modulation target in this example.

The modulation source—LFO1—is used to modulate the frequency, or pitch, of all three oscillators (Pitch123).

You'll hear a vibrato—a modulation of the pitch—at a speed determined by LFO 1's Rate parameter.

The modulation intensity is controlled by the modulation wheel, the range of which is determined by the via parameter. This provides you with control over the depth of vibrato—pitch modulation—via the modulation wheel of your keyboard. This type of configuration is used in countless sound settings.

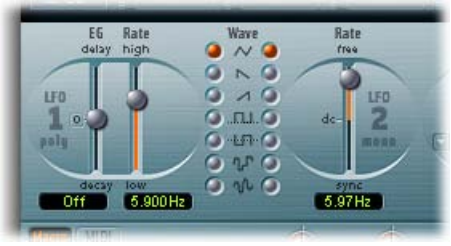
Getting to Know the ES2 LFOs

The ES2 features two multiwaveform LFOs. Both are available as *sources* in the router.

LFO 1 is polyphonic, which means that if it is used for any modulation of multiple voices, they will *not* be phase-locked. Furthermore, LFO 1 is key-synced: Each time you play a key, the LFO 1 modulation of this voice is started from zero.

- To understand the nonphase-locked characteristic more fully, imagine a scenario where a chord is played on the keyboard. If LFO 1 is used to modulate pitch, for example, the pitch of one voice may rise, the pitch of another voice might fall, and the pitch of a third voice may reach its minimum value. As you can see, the modulation is independent for each voice, or note.
- The key-sync feature ensures that the LFO waveform cycle always starts from zero, which results in consistent modulation of each voice. If the LFO waveform cycles were not synchronized in this way, individual note modulations would be uneven.
- LFO 1 can also be faded in or out automatically, courtesy of a built-in envelope generator.

LFO 2 is monophonic, which means that the modulation is identical for all voices. For example, imagine a scenario where a chord is played on the keyboard. If LFO 2 is used to modulate pitch, the pitch of all voices in the played chord will rise and fall synchronously.



- *LFO 1 EG slider:* Controls the time it takes for the LFO modulation to fade in or fade out (see Using LFO 1's Envelope Generator in the ES2).
- *LFO 1 Rate slider:* Defines the frequency—the speed—of the LFO 1 modulation. The value is displayed in Hertz (Hz) beneath the slider.
- *LFO 1 Wave buttons:* This is where you choose the desired waveform for LFO 1. See Using ES2 LFO Waveforms for details on how to use them.
- *LFO 2 Rate slider:* This parameter defines the frequency—speed—of the LFO 2 modulation. See Setting LFO 2's Rate in the ES2.

A Brief LFO Overview

LFO is an abbreviation for *low frequency oscillator*. As the name suggests, this is an oscillator, much like the three main oscillators of the ES2, but it differs in the following ways:

- An LFO generates signals below the audio frequency range—in the bandwidth that falls between 0.1 and 20 Hz, but sometimes as high as 50 Hz.
- Because it can't be heard, an LFO only serves as a modulation source for periodic, cyclic modulation effects—rather than as part of the actual audio signal.

For further information, see Synthesizer Basics.

Using ES2 LFO Waveforms

The LFO 1 Wave buttons allow you to choose different waveforms for LFO 1. The table below outlines how these can affect your sounds.

Try using the waveforms while a modulation routing of Pitch123 (the pitch of all three oscillators) is engaged and running.

Waveform	Comments
Triangle	Well suited for vibrato effects
Sawtooth	Well suited for helicopter and space gun sounds. Intense modulations of the oscillator frequencies with a negative (inverse) sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance creates rhythmic effects. The waveform can also be inverted, resulting in a different start point for the modulation cycle.
Rectangle	Use of the rectangular waves will periodically switch the LFO between two values. The upper rectangular wave switches between a positive value and zero. The lower wave switches between a positive and a negative value set to the same amount above/below zero. An interesting effect can be achieved by modulating Pitch123 with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.
Sample & Hold	The two lower waveform settings of the LFOs output *random* values. A random value is selected at regular intervals, as defined by the LFO rate. The upper waveform steps between randomized values—rapid switches between values. At its lower setting the random wave is smoothed out, resulting in fluid changes to values. The term Sample & Hold (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then *held* until the next *sample* is taken.*Tip:* A random modulation of Pitch123 leads to an effect commonly referred to as a random pitch pattern generator or sample and hold. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies!

Using LFO 1's Envelope Generator in the ES2

LFO 1 features a simple envelope generator, which is used to control the time it takes for the LFO modulation to fade in or fade out. At its center position, which can be accessed by clicking the middle mark, the modulation intensity is static—that is, no fade in or fade out will occur.

To set the LFO 1 modulation fade time

- Choose a positive LFO 1 EG value to *fade in* the modulation.

  The higher the value, the longer the delay time.

- Choose a negative LFO 1 EG value to *fade out* the modulation.

  The lower the slider is positioned onscreen, the shorter the fade out time.

LFO envelopes are most often used for delayed vibrato—many instrumentalists and singers intonate longer notes this way.

To set up a delayed vibrato

1. Place the LFO 1 EG slider at a position in the upper half (Delay) and modulate the Pitch123 target with the LFO1 source in the router.
2. Set a slight modulation intensity.
3. Select an LFO 1 Rate of about 5 Hz.
4. Choose the triangular wave as the LFO 1 waveform.

Tip: Chaotic and fast modulations of oscillator frequencies (target: Pitch123) by the LFO 1 source—with a delayed Sample&Hold waveform, a high Rate, and short fade out—is ideal for emulating the attack phase of brass instruments.

Setting LFO 2's Rate in the ES2

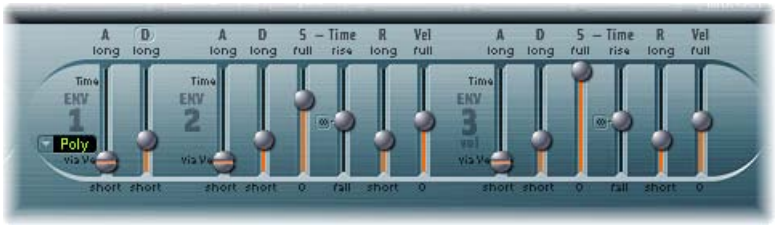
LFO 2 is ideally suited for creating rhythmic modulation effects—that retain perfect synchronicity, even during project tempo changes.

The LFO 2 Rate parameter allows LFO 2 to run freely (in the upper half of the Rate slider range), or to be synchronized with the project tempo (in the lower half of the Rate slider range).

The rate is displayed in Hertz or rhythmic values (the latter when project tempo synchronization is active). Rates range from speeds of 1/64-notes to a periodic duration of 32 bars. Triplet and punctuated values are also available.

Getting to Know the ES2 Envelopes (ENV 1 to ENV 3)

The ES2 features three envelope generators per voice. They are abbreviated as ENV 1, ENV 2, and ENV 3, respectively, in the interface and *router*. In addition, the ES2 features the sophisticated Vector Envelope (see Getting to Know the ES2's Vector Envelope).



Note: To learn more about the roots of the term "envelope generator" and its basic functionality, see Synthesizer Basics.

The parameters of ENV 2 and ENV 3 are identical. ENV 3 defines the changes in level over time for each note played. You can regard ENV 3 as being hard-wired to the router's AMP modulation target.

Envelope Modulation of ES2 Filter Cutoff

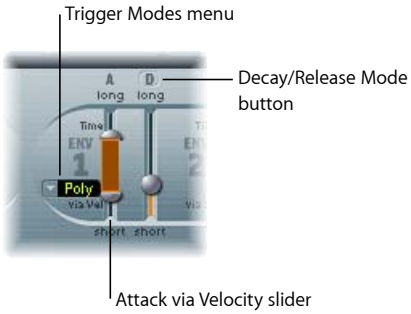
Unlike many other synthesizers, the ES2 has no hard-wired connection between any of the envelope generators and the filter cutoff frequencies.

Set up a modulation routing as follows, in order to establish this type of modulation: Set target to Cutoff 1, Cutoff 2, or Cut 1+2. Set source to ENV 2, for example. After it is set as described, the slider to the right of the modulation routing will function as the filter's EG Depth parameter.

Note: Both ENV 2 and ENV 3 are velocity sensitive, making it unnecessary to set via to Velo in the modulation routing; you can leave via off.

Getting to Know ENV 1 in the ES2

Although ENV 1 might appear to be rather poorly equipped at first glance, its handful of parameters are useful for a vast range of synthesizer functions.



- *Trigger Modes menu:* You can define the trigger behavior of ENV 1 by choosing one of the following settings:
  - *Poly:* The envelope generator behaves as you would expect on any polyphonic synthesizer: Every voice has its own envelope.
  - *Mono:* A single envelope generator modulates all voices in the same way. All notes must be released before the envelope can be retriggered. If you play legato, or any key remains depressed, the envelope won't restart its attack phase.
  - *Retrig:* A single envelope generator modulates all voices in the same way. The envelope will be triggered by any key you strike, even when other notes are sustained. All sustained notes are identically affected by the retriggered envelope.
- *Attack via Velocity slider:* The Attack time slider is divided into two halves. The lower half defines the attack time when the keys are struck hard—at maximum velocity. The upper half defines the attack time at minimum velocity. Drag the area between the two slider halves to move them both simultaneously. If this area is too small to drag, click an unused portion of the slider, and drag vertically.
- *Decay/Release mode button:* Click to switch ENV 1 between behaving as an Attack/Decay and an Attack/Release envelope.

Setting Envelope 1 Decay or Release in the ES2

ENV 1 can be set to act as an envelope generator with either of the following: an Attack time and Decay time parameter *or* an Attack time and Release time parameter.

Click here to switch between Decay and Release modes.

To switch between Attack/Decay and Attack/Release modes

- Click the D or the R above the right ENV 1 slider. The button label will change to reflect the mode that is activated.
  - *In Attack/Decay mode:* The level will fall to zero after the attack phase has completed, whether or not the note is sustained. It will decay at the same speed—even if you release the key. The decay time is set with the D (Decay time) slider.
  - *In Attack/Release mode:* The envelope level remains at its maximum after the attack phase is over, while the key remains depressed. Following the release of the key, the level decreases over the time period defined by the R (Release time) slider.

Emulating Classic Polysynth Filter Behavior in the ES2

Early analog polysynths passed all voices through a single lowpass filter. This design decision was primarily due to cost factors. The best-known examples of these polyphonic instruments were the Moog Polymoog, the Yamaha SK20, and the Korg Poly-800. The sole lowpass filter of such instruments is controlled by a single envelope generator. To simulate this behavior in the ES2, use the Mono or Retrigger modes.

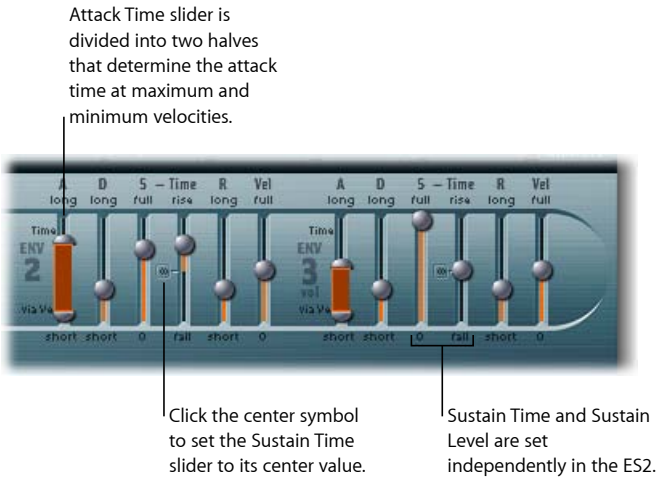
Modulation of the Cutoff 2 target with a percussive source, such as ENV 1, which is set to Retrig will have a percussive filter effect every time you play another key, if you play and sustain a bass note. The newly struck key is also shaped by the same filter. Playing a sound set up in this way feels like you're playing a polyphonic synthesizer with *one* filter. This is despite the fact that the ES2's filters remain polyphonic and can be simultaneously modulated by different polyphonic sources.

As a further tip, you can also simulate the percussion of a Hammond organ by using the Mono or Retrigger modes.

Getting to Know ENV 2 and ENV 3 in the ES2

The feature sets of ENV 2 and ENV 3 are identical, but it is always the task of ENV 3 to define the level of each note—to modulate the dynamic stage, in other words.

ENV 3 is, however, also available for simultaneous use as a source in the router. The envelope time parameters can also be used as modulation targets in the router.


Attack Time slider is divided into two halves that determine the attack time at maximum and minimum velocities.
A long
D long
S full
Time rise
R long
Vel full
ENV 2
ENV 3
short
fall
Click the center symbol to set the Sustain Time slider to its center value.
Sustain Time and Sustain Level are set independently in the ES2.

- *Attack slider:* Defines the length of time it takes for the level of a note to rise from an amplitude of zero to the set amplitude. The Attack time sliders of ENV 2 and ENV 3 are divided into two halves.
  - The lower half defines the attack time when the keys are struck hard, at maximum velocity. The upper half defines the attack time at minimum velocity. Drag the area between the two slider halves to move them both simultaneously. If this area is too small to be dragged, click in an unused portion of the slider and drag vertically.
- *Decay slider:* Determines the period it takes for the level of a held note to fall to the sustain level, after the attack phase has completed.
  - If the Sustain level parameter is set to its maximum value, the Decay parameter has no effect.
  - When the Sustain level is set to its minimum value, the Decay parameter defines the duration or fade-out time of the note.
- *Sustain and Sustain Time sliders:* There are two sustain parameters that interact with each other. One controls the sustain level, and the other controls the sustain time. See Using the Envelope 2 and 3 Sustain Parameters in the ES2.
- *(R) Release Time slider:* Defines the time required for the (sustain) level to decay to zero—after the key is released.
- *Vel (Velocity Sensitivity) slider:* Determines the velocity sensitivity of the entire envelope. If set to maximum, the envelope will output its maximum level only when the keys are struck at maximum velocity. Softer velocities result in a corresponding change to the envelope levels—with a 50% velocity resulting in half-levels for each envelope-level parameter.

Using the Envelope 2 and 3 Sustain Parameters in the ES2

When the Sustain Time (rise) slider is set to its center value, the Sustain (S) Level slider behaves like the sustain parameter of any synthesizer ADSR envelope.

In this position, the Sustain (Level) slider defines the level that is sustained while the key remains depressed, following completion of the Attack time and Decay time phases.

The Sustain Time slider defines the time it takes for the level to rise from the Sustain level back to its maximum level—or to fall to zero:

- Settings in the lower half of the Sustain Time slider range (fall) determine the time required for the level to decay from the sustain level to zero. The lower the slider position, the faster the sound level decays.
- Settings in the upper half of its range (rise) determine the time required for the level to rise from the sustain level to its maximum value. The higher the slider position, the faster the sound level rises.

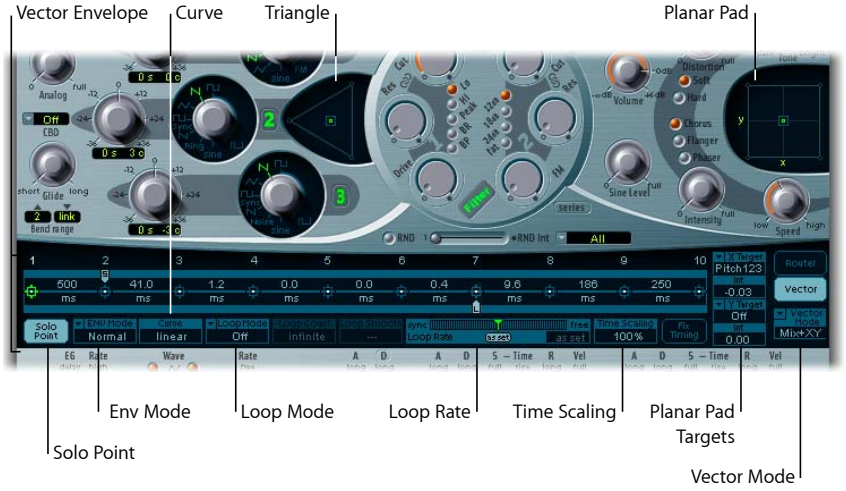
Emulating Instrument Behaviors with Envelope Modulation

On pianos and plucked string instruments, high notes decay faster than low notes.

To simulate this effect, modulate the Decay Time target with the Kybd source in the router. The modulation intensity slider should be set to a negative value.

Getting to Know the ES2's Vector Envelope

The Vector Envelope is a multipoint, loop-capable control source. Its sole purpose is to provide real-time control of square icon movements in the Triangle and the Planar Pad. The Vector Envelope shares the space occupied by the modulation router, and it can be viewed by clicking the Vector Envelope button to the right of the router.



Each voice is equipped with an independent Vector Envelope, which is triggered from its start point with every new keystrike (MIDI note-on message).

Conceptually, the Vector Envelope—and Planar Pad and Triangle—may seem somewhat strange, and perhaps a little intimidating, but a little experimentation on your part will reveal how easy these features are to use. Combining these facilities with other ES2 synthesis options enables you to create some truly unique sounds that are—quite literally—moving.

To activate or deactivate the Vector Envelope

- Turn off the Solo Point button (described on Setting the ES2 Vector Envelope Solo Point) to activate the Vector Envelope.
- Turn on the Solo Point button to deactivate the Vector Envelope.

When Solo Point is turned on, only the currently selected Triangle and Planar Pad positions of the currently selected point are active.

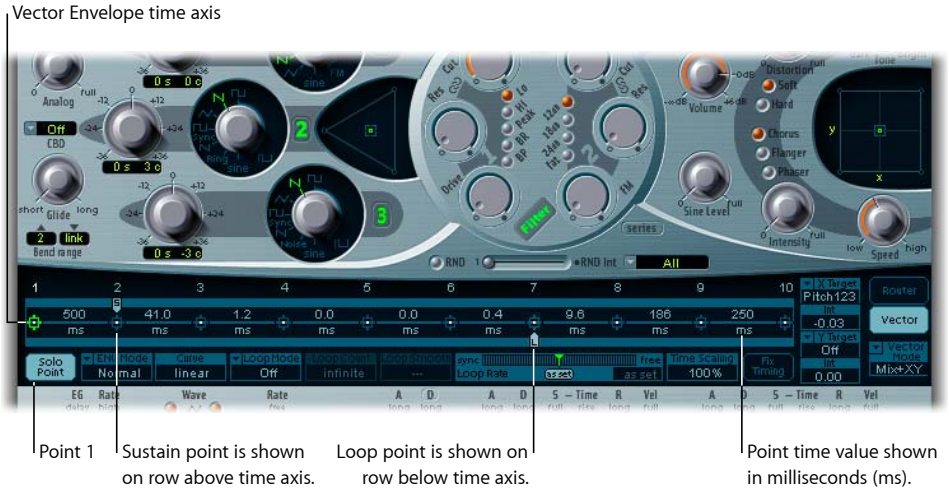
Vector Envelope Control of the ES2's Planar Pad and Triangle

The Vector Mode pop-up menu below the Vector Envelope button allows you to specify the target for the Vector Envelope—the Planar Pad and/or the Triangle.

- *Off:* The Vector Envelope does not control the Triangle or the Planar Pad. It is completely turned off. This allows you to manually set and control the square icons of the Triangle and the Planar Pad.
- *Mix:* The Vector Envelope controls the Triangle, but not the Planar Pad.
- *XY:* The Vector Envelope controls the Planar Pad, but not the Triangle.
- *Mix+XY:* The Vector Envelope controls both the Planar Pad and the Triangle.

An Overview of ES2 Vector Envelope Points, Times, and Loops

The Vector Envelope time axis runs from left to right.


Vector Envelope time axis
Point 1
Sustain point is shown on row above time axis.
Loop point is shown on row below time axis.
Point time value shown in milliseconds (ms).

Up to 16 points can be displayed on the time axis (10 are shown in the figure above). Each point can control the square icon positions of the Triangle and the Planar Pad (as described in Vector Envelope Control of the ES2's Planar Pad and Triangle).

The points are numbered sequentially—from left to right—along the time axis.

There are always at least three points: Point 1 is the start point, point 2 is defined as the Sustain point, and point three is the end point.

Any point can be declared the *Sustain point*. If a played note is held for a sufficient length of time and there's no loop engaged, any envelope movement will stop when the Sustain point is reached. The Sustain point value is maintained until the key is released—until the MIDI note-off command.

Any point can be declared the *Loop point*. The looped area spans the time between the Sustain point and Loop point. In between these points you can create further points that describe the movements of the square icons in the Planar Pad and Triangle.

Selecting, Creating, and Deleting ES2 Vector Envelope Points

The more points you set, the more complex the movements that can be performed.

To select a point

- Click to select. Once selected, you can freely edit the point.

To create a new point

- Shift-click between two existing points.

The segment that previously existed between the two old points is divided at the clicked position. The sum of the two new segment times is equal to the time of the original undivided segment. This ensures that following points retain their absolute time positions. Existing square icon positions in the Triangle and Planar Pad are fixed, thus ensuring that newly created points don't affect any previously defined movements.

To delete a point

- Control-click it.

Reverting to Default ES2 Vector Envelope Point Values

On occasion you may want to revert to the default values of a point. This is done directly in the Triangle or the Planar Pad:

To revert to the default value for a point, do one of the following

- Option-click the Triangle. The square icon is set to the center position of the Triangle.

  All oscillators are set to output the same level.

- Option-click the Planar Pad.

  The square icon is set to the center position of the Planar Pad. Both axis values are set to zero.

Setting the ES2 Vector Envelope Solo Point

The Solo Point button basically turns off the Vector Envelope. If the Solo Point button is activated, no dynamic modulations are generated by the Vector Envelope. In this scenario, the currently visible square icon positions of the Triangle and Planar Pad are permanently in effect. These square icon positions match the currently selected Vector Envelope point.

If you select another Vector Envelope point by clicking it, the square icon positions in the Triangle and Planar Pad will update to reflect your selection. If the Solo Point button is turned on, the newly selected point will become the Solo point.

Note: You can independently switch off Vector Envelope modulation of the Planar Pad by setting Vector Mode to off, as described in Vector Envelope Control of the ES2's Planar Pad and Triangle.

Setting the ES2 Vector Envelope Sustain Point

As mentioned earlier, any point can be declared the Sustain point. Assuming that the played note is held long enough and there's no loop engaged, any envelope movement will stop when this Sustain point is reached. The Sustain point value is maintained until the key is released—until the MIDI note-off command.

To define a point as the Sustain point

- Click in the turquoise strip above the desired point.

The Sustain point is indicated by an S between the point and its number shown on the turquoise strip.

Setting Up ES2 Vector Envelope Loops

The Vector Envelope can run in one-shot mode—as long as the note is sustained; it can be set to repeat a specific number of times; or it can repeat indefinitely—much like an LFO modulation. You can achieve repetitions by using the *loop functions*.

Although the loop parameters may remind you of the loop parameters available for samples, there are significant differences between them. The Vector Envelope only supplies control signals that are used to move the square icon positions of the Triangle and Planar Pad. The audio output of the ES2 is *not* looped in any way.

Setting the Loop Point

Any point can be declared the Loop point. Provided that the note is held for a suitable length of time, portions of the envelope can be repeated, or looped.

The looped area spans the time between the Sustain point and the Loop point. In-between these points you can define several points that describe square icon movements in the Triangle and the Planar Pad.

To define a point as the Loop point

- Click in the turquoise strip below the desired point.

  A Loop point is indicated by an *L* in the strip below.

Note: To see and define the Loop point, Loop Mode must be activated (see "Setting the Vector Envelope Loop Mode" below).

Setting the ES2 Vector Envelope Loop Mode

You can choose from the following Vector Envelope Loop modes: Off, Forward, Backward, and Alternate.

Click here to choose a Loop mode.

- *Off:* When Loop Mode is set to Off, the Vector Envelope runs in one-shot mode from beginning to end—if the note is held long enough to complete all envelope phases. The other loop parameters are disabled.
- *Forward:* When Loop Mode is set to Forward, the Vector Envelope runs from the beginning to the Sustain point—then begins to periodically repeat the section between the Sustain point and the Loop point—in a forward direction.
- *Backward:* When Loop Mode is set to Backward, the Vector Envelope runs from the beginning to the Sustain point—then begins to periodically repeat the section between the Sustain point and the Loop point—in a backward direction.
- *Alternate:* When Loop Mode is set to Alternate, the Vector Envelope runs from the beginning to the Sustain point—then periodically switches to the Loop point, then back to the Sustain point—alternating between backward and forward directions.

Setting the ES2 Vector Envelope Loop Rate

Just as every LFO has a speed, or *rate*, parameter, the loop can be set to cycle at a defined speed. The Vector Envelope Loop Rate can also be synchronized with the project tempo.

To adjust the Vector Envelope loop rate, do one of the following

- Drag the green indicator in the center of the Loop Rate bar to the left or right.
- Drag vertically in the value field "as set" (shown in the figure below).

- *As set:* If you switch the Loop Rate to "as set," the loop cycle length will equal the sum of the times between the sustain and Loop points. Click the field labeled "as set" below the Rate slider to select it.
- *Rhythmic:* If you set the Loop Rate to one of the rhythmic values (sync) by dragging the Loop Rate indicator towards the left half of the slider, the loop rate will follow the project tempo. You can choose from 32 bars up to a 64th triplet note value.

- *Free:* You can also set a free Loop Rate by dragging the Loop Rate indicator towards the right half of the slider (free). The value indicates the number of cycles per second.

  Note: If Loop Rate is not switched to "as set," and Loop Mode (Forward, Backward, or Alternate) is active, the times of points between the Loop and Sustain points and the Loop Smooth value are shown as a percentage of the loop duration, rather than in milliseconds.

Making Smooth ES2 Vector Envelope Loop Transitions

When Loop Mode is set to Forward or Backward, there will inevitably be a moment when a transition from the Sustain point to the Loop point occurs. You can use the Loop Smooth parameter to even out the transition, thus avoiding abrupt position changes.

- If the Loop Rate parameter is set to Sync or Free, the loop smoothing time is displayed as a percentage of the loop cycle duration.
- If the Loop Rate parameter is "as set," the loop smoothing time is displayed in milliseconds (ms).

Specifying an ES2 Vector Envelope Loop Count

The loop cycle of the Vector Envelope can be repeated a specified number of times. After the defined number of repetitions have occurred, the Vector Envelope will run from the Sustain point onwards. Possible values are 1 to 10 and "infinite."

Setting the ES2 Vector Envelope Release Phase Behavior

There are two release phase options in the Env Mode menu: Normal and Finish.

Using the Normal ES2 Vector Envelope Mode

If the Env Mode menu is set to Normal, the release phase—the phase after the Sustain point—begins as soon as you release the key (note off). In other words, the release phase will start from the Vector Envelope point where you released the key.

The following behaviors apply:

- If looping is turned off and the Vector Envelope reaches the Sustain point, the Sustain point value will be retained for as long as you hold a key.
- If looping is turned on and the Loop point is positioned before the Sustain point, the loop will cycle for as long as you hold a key.
- If looping is turned on and the Loop point is positioned after the Sustain point, the Vector Envelope loop will continue to cycle until the overall release phase of the sound, as determined by the ENV 3 Release parameter, has completed.

Using the Finish ES2 Vector Envelope Mode

If the Env Mode menu is set to Finish, the Vector Envelope does not immediately commence the release phase when you release the key. Rather, it plays all points for their full duration until the end point is reached, regardless of whether you hold the key or release it.

The following behaviors apply:

- If looping is turned off, the Sustain point is ignored. The Vector Envelope will complete all points up to the end point, regardless of whether you hold the key or release it.
- If looping is turned on, the Vector Envelope will play all points until it reaches the Loop point, and then play the loop until the end point is reached. It does not matter if the Loop point is positioned before or after the Sustain point.
- If looping is turned on, and Loop Count is set to a value other than "infinite," the Vector Envelope will continue on to the subsequent points—following completion of the specified number of loop repetitions. If Loop Count is set to "infinite," the points after the loop are irrelevant. See ES2.

Choosing ES2 Vector Envelope Point Transition Shapes

The Curve parameter defines the shape of the transition from point to point. You can choose from nine convex and nine concave shapes. There are also two unusual curve forms, "hold+step" and "step+hold," which allow stepped modulations.

- *step+hold:* This curve jumps at the beginning of the transition.
- *hold+step:* This curve jumps at the end of the transition.

Note: You can use "hold+step" to create stepped vector grooves—with up to 15 steps.

Setting ES2 Vector Envelope Times

With the exception of the first point, which is tied to the beginning of each played note, every point has a Time parameter. This parameter defines the period of time required for the position indicator to travel from the point that preceded it. The times are normally displayed in milliseconds (ms).

To adjust a time value

- Click the numerical value and drag vertically.

Note: Changing a time value alters the absolute time positions of all subsequent points.

To adjust a time value without affecting the absolute time positions of later points

- Control-drag the Time parameter to increase or decrease the time required to reach the following point. The time setting of the ensuing point is simultaneously adjusted—by a corresponding amount. This ensures that the adjacent and all following points retain their absolute time positions.

Time Scaling the ES2's Vector Envelope

You can stretch and compress the entire Vector Envelope. This is extremely useful if you want to double the Vector Envelope's speed, for example. Rather than halving the time values of every point, you can set the Time Scaling parameter above the ENV3 Attack slider to 50%.

Click here to change the time scale.

- The Time Scaling parameter ranges from 10% to 1000%. It is scaled logarithmically.
- If the Loop Rate is "as set," scaling also affects the loop.
- If the Loop Rate is set to a free or synced value, the setting is not affected by the Time Scaling parameter.

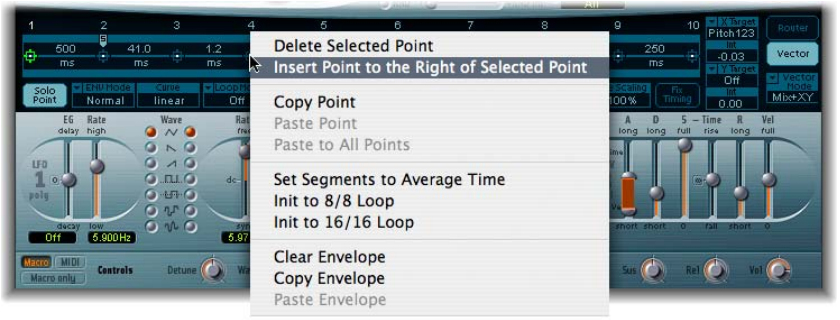
ES2 Fix Timing—Normalizing Time Scaling and the Loop Rate

Clicking the Fix Timing button (to the right of the Time Scaling parameter) multiplies the Time Scaling value by all time parameters, and Time Scaling will be reset to 100%. There will be no audible difference. This is simply a normalizing procedure, similar to the Normalizing function in the Region parameter box.

In cases where Loop Rate is set to a synced value, clicking Fix Timing will switch the Loop Rate to "as set," thus preserving the absolute rate.

Using the ES2 Vector Envelope Shortcut Menu

A number of Vector Envelope commands and functions can be accessed by right-clicking anywhere in the Vector Envelope. This will launch the pop-up menu shown below.



Choose any item in the menu to perform the command or function.

Using the ES2's Planar Pad

The Planar Pad has two axes—X and Y. The X axis is on the horizontal plane and the Y axis is on the vertical plane.

You can modulate one user-defined parameter with the X value and another user-defined parameter with the Y value, allowing you to use the mouse like a joystick.

The X and Y axes have positive and negative value ranges. When you drag the square icon, the values of both axes are continuously transmitted.

Choosing a Vector Target—Modulation Destinations

The Vector X and Vector Y Target menus determine which parameter is modulated by square icon movements in the Planar Pad. The modulation targets are identical to those in the router. See ES2 Modulation Target Reference for descriptions.

The position of the square icon in the Planar Pad is also available in the router, as the Pad-X and Pad-Y source and via options. See ES2 Modulation Source Reference and Using Via Sources to Control ES2 Modulation Intensity.

Setting the Vector Intensity—Defining the Modulation Intensity

The maximum intensity, sensitivity, and polarity of the modulation is set with the Vector X Int and Vector Y Int parameters.

To adjust the modulation intensity

- Drag vertically in the Vector X and Y Int fields.

Use a negative value to invert the modulation polarity.

ES2 Modulation Target Reference

The following targets are available for real-time modulation.

ES2 Oscillator Targets

The table below includes all oscillator-related modulation targets.

Target	Comments
Pitch123	Modulates the frequencies (pitch) of all three oscillators. If you select an LFO as the source, this target leads to siren or vibrato sounds. Select one of the envelope generators with zero attack, short decay, zero sustain, and short release as the source for tom and kick drum sounds.
Pitch 1	Modulates the frequency (pitch) of Oscillator 1. Slight envelope modulations can make the amount of detuning change over time, when Oscillator 1 is sounding in unison with another (unmodulated) oscillator. This also applies to the other Pitch targets and is particularly useful for synthesizer brass sounds.
Pitch 2	Modulates the frequency (pitch) of Oscillator 2.
Pitch 3	Modulates the frequency (pitch) of Oscillator 3.

Target	Comments
Detune	Controls the amount of detuning between all three oscillators. The sensitivity of all pitch modulation targets is determined by the modulation intensity. This is scaled as per the lists below, allowing you to create very delicate vibrati in the cent range (1/100 semitone), and huge pitch jumps by octaves. • Modulation intensity from 0 to 8: steps are 1.25 cents. • Modulation intensity from 8 to 20: steps are 3.33 cents. • Modulation intensity from 20 to 28: steps are 6.25 cents. • Modulation intensity from 28 to 36: steps are 12.5 cents. • Modulation intensity from 36 to 76: steps are 25 cents. • Modulation intensity from 76 to 100: steps are 100 cents. This leads to the following rules of thumb for modulation intensity values. • Intensity of 8 equals a pitch shift of 10 cents. • Intensity of 20 equals a pitch shift of 50 cents (one quarter tone). • Intensity of 28 equals a pitch shift of 100 cents (one semitone). • Intensity of 36 equals a pitch shift of 200 cents (two semitones). • Intensity of 76 equals a pitch shift of 1,200 cents (one octave). • Intensity of 100 equals a pitch shift of 3,600 cents (three octaves).
OscWaves	Depending on the waveforms set in the three oscillators, this target can be used to modulate: • The pulse width of rectangular and pulse waves • The amount of frequency modulation (Oscillator 1 only) • Noise color (Oscillator 3 only) • The position of the Digiwaves OscWaves affects all oscillators simultaneously. For further information about the effects of these modulations, see Using Pulse Width Modulation in the ES2, Using Frequency Modulation in the ES2, Using Noise in the ES2 (Oscillator 3 Only), and Using Digiwaves in the ES2.
Osc1Wave	Depending on the *waveform* selected for Oscillator 1, you can control the pulse width of rectangular and pulse waves, the amount of frequency modulation, or the position of the Digiwave. *Note:* In classic FM synthesizers the amount of FM is controlled in real time by velocity-sensitive envelope generators. Select one of the ENVs as the source for such sounds.
Osc2Wave	The same as Osc1Wave, except that Oscillator 2 does not feature FM. Note that pulse width modulation also works with both the synchronized rectangular and ring modulated rectangular waves.
Osc3Wave	Oscillator 3 is the same as Osc1Wave and Osc2Wave except that it does not feature FM or ring modulation. Oscillator 3 features noise, the color of which can be modulated with this parameter.

Target	Comments
OscWaveB	The transitions between Digiwaves during a wavetable modulation (where you switch between different Digiwaves) are always smooth. You can use the OscWaveB target to continuously modulate the shape of the transitions from smooth to hard. This target applies to all oscillators.
Osc1WaveB	If wavetable modulation is active for a Digiwave (using the Osc1Wav target), you can use this target to modulate the shape of the transition. When you are frequency-modulating Oscillator 1, the Osc1WaveB target offers much higher FM intensities than either the Osc1 FM or the Osc1Wave targets.
Osc2WaveB	The same as above for a Digiwave using the Osc2Wav target.
Osc3WaveB	The same as above for a Digiwave using the Osc3Wav target.
SineLev1	SineLevl (Sine Level) allows the *sine wave* level of Oscillator 1 to be modulated. The parameter defines the level of the first partial tone of Oscillator 1. See Thickening the ES2's Sound with Sine Level.
OscLScle	OscLScle (Osc Level Scale) modulates the levels of *all three* oscillators simultaneously. A modulation value of *0* mutes all oscillators, whereas a value of *1* raises the gain of the entire mix by 12 dB. The modulation is applied *before* the overdrive stage, allowing for dynamic distortions.
Osc1Levl	(Osc 1 Level) allows modulation of Oscillator 1's level.
Osc2Levl	(Osc 2 Level) allows modulation of Oscillator 2's level.
Osc3Levl	(Osc 3 Level) allows modulation of Oscillator 3's level.

ES2 Filter Targets

The table below includes all filter-related modulation targets.

Target	Comments
Cutoff 1	Modulates the Cutoff Frequency parameter of Filter 1. See Using the ES2 Filter Cutoff and Resonance Parameters.
Resonance 1	(Reso 1) Modulates the Resonance parameter of Filter 1.
Cutoff 2	Modulates the Cutoff Frequency parameter of Filter 2.
Resonance 2	(Reso 2) Modulates the Resonance parameter of Filter 2.
LPF FM	Determines the intensity of the lowpass filter frequency modulation (LPF FM) of Filter 2—with a sine wave (at the same frequency as Oscillator 1). This parameter is described in Modulating Filter 2's Frequency in the ES2.
Cut 1+2	Modulates the cutoff frequency of both filters in parallel. This is like applying the same modulation to Cutoff 1 and Cutoff 2 in two *modulation routings*.

Target	Comments
Cut1inv2	Cut1inv2 (Cutoff 1 normal and Cutoff 2 inverse) simultaneously modulates the *cutoff* frequencies of the first and second filters *inversely* (in opposite directions). Put another way, when the first filter's *cutoff* frequency is rising, the *cutoff* of the second filter will fall—and vice versa. In cases where you have combined Filter 1, defined as a highpass filter, and Filter 2 in *serial* mode, both act as a bandpass filter. Modulating the *Cut1 inv 2target* will result in a modulation of the bandpass filter's bandwidth in this scenario.
Filter Blend	(FltBlend) modulates the Filter Blend parameter. See Getting to Know the ES2 interface.

Other ES2 Targets

The table below includes all other modulation targets.

Target	Comments
Amp	This target modulates the dynamic stage, or level of voices. If you select Amp as the target and modulate it with an LFO as the source, the level will change periodically, and you will hear a tremolo.
Pan	This *target* modulates the panorama position of the sound in the stereo spectrum. Modulating Pan with an LFO will result in a stereo tremolo (auto panning). In unison mode, the panorama positions of all voices are spread across the entire stereo spectrum. Nevertheless, pan can still be modulated, with positions being moved in parallel.
LFO1Asym	(LFO1 Asymmetry) can modulate the selected waveform of LFO 1. If a square wave, it changes pulse width. If a triangle wave, it sweeps between triangle and sawtooth. If a sawtooth wave, it shifts its zero crossing.
LFO1Curve	This *target* modulates the waveform smoothing of the square and random wave. If the LFO is using a triangle or sawtooth wave, it changes between convex, linear, and concave curves.

Scaled ES2 Modulation Targets

All of the following modulation targets result in a scaled modulation, which means that the target parameter value will be multiplied by the modulation value. This works as follows: a modulation value of 0.0 results in no change, a modulation value of +1.0 equals a 10x multiplication, and a modulation value of –1.0 equals a multiplication by 0.04.

Target	Comments
LFO1Rate	This *target* modulates the frequency (rate) of LFO 1. You can automatically accelerate or slow down LFO 1's rate by modulating the LFO1Rate target with one of the envelope generators (ENV) or with LFO2.
Env2Atck	(Envelope 2 Attack) modulates the Attack time of the second envelope generator.

Target	Comments
Env2Dec	(Envelope 2 Decay) modulates the Decay time of the second envelope generator. In cases where you've selected Env2Dec as the target and Velocity as the source, the duration of the decaying note is dependent on how hard you strike the key. Selecting Keyboard as the source will result in higher notes decaying more quickly (or slowly).
Env2Rel	Env2Rel (Envelope 2 Release) modulates the Release time of the second envelope generator.
Env2Time	Env2Time (Envelope 2 All Times) modulates *all* of ENV2's time parameters: Attack, Decay, Sustain, and Release times.
Env3Atck	Env3Atck (Envelope 3 Attack) modulates the Attack time of ENV3.
Env3Dec	Env3Dec (Envelope 3 Decay) modulates the Decay time of ENV3.
Env3Rel	Env3Rel (Envelope 3 Release) modulates the Release time of ENV3.
Env3Time	Env3Time (Envelope 3 All Times) modulates all ENV3 time parameters: Attack, Decay, Sustain, and Release times.
Glide	This target modulates the duration of the Glide (portamento) effect. If you modulate Glide, with Velocity selected as the source, the speed of the keystrike determines the time it takes for the played notes to reach the target pitch.

ES2 Modulation Source Reference

The following modulation sources are available:

Source	Comment
LFO1	LFO 1 is used as a source.
LFO2	LFO 2 is used as a source.
ENV1	Envelope Generator 1 is used as a source.
ENV2	Envelope Generator 2 is used as a source.
ENV3	Envelope Generator 3 is used as a source. Envelope Generator 3 always controls the level of the overall sound.
Pad-X, Pad-Y	Define the axes of the Planar Pad as modulation sources for the selected modulation target. See Using the ES2's Planar Pad and Getting to Know the ES2's Vector Envelope.
Max	Max sets the value of this *source* to +1. This offers interesting options for controlling the modulation intensity with all possible *via* values.
Kybd	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of –1 or +1, respectively is sent. Modulate the Cut 1+2 target with the Kybd source to control the cutoff frequencies of the filters with the keyboard position—as you play up and down the keyboard, the cutoff frequencies change. A modulation intensity of 0.5 proportionately scales cutoff frequencies with keyboard note pitches.

Source	Comment
Velo	Velocity sensitivity serves as a modulation *source*.
Bender	The pitch bend wheel serves as a bipolar modulation source. This is also true when the Bend Range parameter of the oscillators is set to 0.
ModWhl	The modulation wheel serves as a modulation source.*Note:* For most standard applications, you'll probably use the wheel as the *via* controller. Traditionally, it is used to control the intensity of periodic LFO modulations. Used here, it can be employed for direct, static modulations, such as controlling both filter cutoff frequencies (Target = Cut 1+2).
Touch	Aftertouch serves as a modulation source. The ES2 reacts to poly pressure (polyphonic aftertouch).*Note:* If you set the *Target* to Cut 1+2, the cutoff frequencies will rise and fall, depending on how firmly you press a key on your touch-sensitive MIDI keyboard—after the initial keystrike.
Whl+To	Both the modulation wheel *and* aftertouch serve as modulation sources.
MIDI Controllers A-F	MIDI controllers available in the router are named Ctrl A–F and can be assigned to arbitrary controller numbers. Please see the Using Macro Controls and Assigning Controllers in the ES2.
RndN01	RndNO1 (Note On Random1) outputs a random modulation value between −1.0 and 1.0, that changes when a note is triggered or retriggered. The (random) note-on modulation remains constant throughout the note duration, until the next note-on trigger.*Note:* There is no value change when playing legato while in legato mode.
RndN02	RndNO2 (Note On Random 2) behaves like Note On Random1, but it glides, rather than steps, to the new random value—using the Glide time (inclusive of modulation). It also differs from Note On Random 1 in that the random modulation value changes when playing legato while in legato mode.
SideCh	SideCh (Side Chain modulation) uses a side chain signal as a modulation (trigger) signal. The side chain source can be selected in the Side Chain menu in the upper gray area of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.

ES2 Modulation Via Source Reference

The following sources may be used to control the modulation intensity.

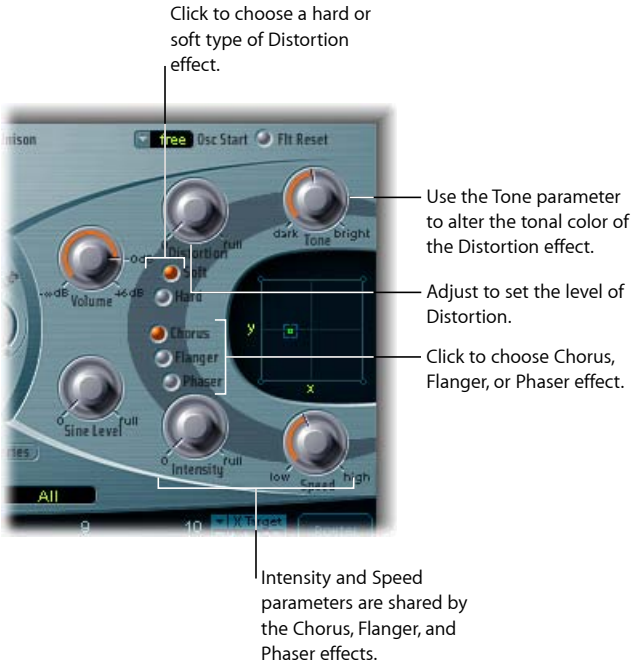
Via source	Comment
LFO1	The modulation undulates at the speed and waveform of LFO 1, which controls the modulation intensity.
LFO2	The modulation undulates at the speed and waveform of LFO 2, which controls the modulation intensity.

Via source	Comment
ENV1	ENV1 controls the modulation intensity.
ENV2	ENV2 controls the modulation intensity.
ENV3	ENV3 controls the modulation intensity.
Pad-X, Pad-Y	Both axes of the Planar Pad are available as *viasources* as well, enabling you to control modulation intensities with them.
Kybd	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of *0*). Five octaves below and above, an output value of –1 or +1, respectively, is sent. If you select Pitch123 as the target, modulate it with the LFO1 source, and select Keyboard as the via value, the vibrato depth will change, depending on the key position. Put another way, the vibrato depth will be different for notes higher or lower than the defined Keyboard position.
Velo	If you select Velo (Velocity) as the via value, the modulation intensity will be velocity sensitive—modulation will be more or less intense depending on how quickly (how hard) you strike the key.
Bender	The pitch bend wheel controls the modulation intensity.
ModWhl	If you select ModWhl (Modulation Wheel) as the via value, the modulation intensity will be controlled by your MIDI keyboard's modulation wheel.
Touch	If you select Touch (Aftertouch) as the via value, the modulation intensity will be touch sensitive—modulation will be more or less intense depending on how firmly you press the key of your touch-sensitive MIDI keyboard after the initial keystrike (aftertouch is also known as *pressure sensitivity*).
Whl+To	Both the modulation wheel and aftertouch control the modulation intensity.
MIDI Controllers A-F	MIDI controllers available in the router are named Ctrl A–F, rather than Expression, Breath, and General Purpose 1–4 (MIDI Control Change Messages 16 to 19 are also known as General Purpose Slider 1/2/3/4). These can be assigned to arbitrary controller numbers via menus in the Controller Assignments section at the bottom of the interface (press the MIDI button to view menus A to F).
RndN01	RndNO1 (Note On Random1) outputs a random modulation intensity value between –1.0 and 1.0, which changes when a note is triggered or retriggered. The random note-on modulation remains constant throughout the note duration, until the next note-on trigger.*Note:* There is no value change when playing legato while in legato mode.
RndN02	RndNO2 (Note On Random 2) behaves like Note On Random1, but it glides, rather than steps, to the new random intensity value—using the Glide time (inclusive of modulation). It also differs from Note On Random 1 in that the random modulation value changes when playing legato while in legato mode.

Via source	Comment
SideCh	SideCh (Side Chain modulation) uses a side chain signal as a modulation intensity (trigger) signal. The side chain source can be selected in the Side Chain menu in the upper gray area of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.

Using the ES2's Integrated Effect Processing Section

The ES2 is equipped with an integrated effect processor. Any changes to the parameters of these effects are saved with each sound setting.



You can activate only two effects at the same time.

- Distortion
- A choice of the Chorus, Flanger or Phaser effect. These effects share the same control knobs—Intensity and Speed.

ES2 Distortion Effect

The Distortion effect offers the following parameters:

- *Soft button:* Activates the Distortion effect Soft mode. The distortion circuit sounds somewhat like a tube overdrive.

- *Hard button:* Activates the Distortion effect Hard mode. The distortion effect sounds like a fully transistorized fuzz box.
- *Distortion knob:* Defines the amount of distortion. Turn this knob to zero to disable the effect.
- *Tone knob:* Controls the treble portion of the distortion effect output.

ES2 Chorus Effect

A *chorus* effect is based on a delay line, the output of which is mixed with the original, dry signal. The short delay time is modulated periodically, resulting in pitch deviations. The modulated deviations, in conjunction with the original signal's pitch, produce the chorus effect.

To activate the Chorus effect

- Enable the Chorus button.
  - The Intensity parameter determines the depth of the effect (how "rich" the modulation is). Turn this knob to zero to disable the effect.
  - The Speed parameter determines the modulation rate.

ES2 Flanger Effect

A *flanger* works in a similar fashion to a chorus, but with even shorter delay times. The output signal is fed back into the input of the delay line. This feedback results in the creation of harmonic resonances—which wander cyclically through the spectrum, giving the signal a "metallic" sound.

To activate the Flanger effect

- Click the Flanger button.
  - The Intensity parameter determines the depth of the effect—how "cutting" the modulation is. Turn this knob to zero to disable the effect.
  - The Speed parameter determines the modulation rate.

ES2 Phaser Effect

A *phaser* mixes a delayed and an original signal. The delayed element is derived from an allpass filter, which applies a frequency-dependent delay to the signal. This is expressed as a phase angle. The effect is based on a comb filter, which is basically an array of inharmonic notches—rather than resonances, as with the flanger—which also wander through the frequency spectrum.

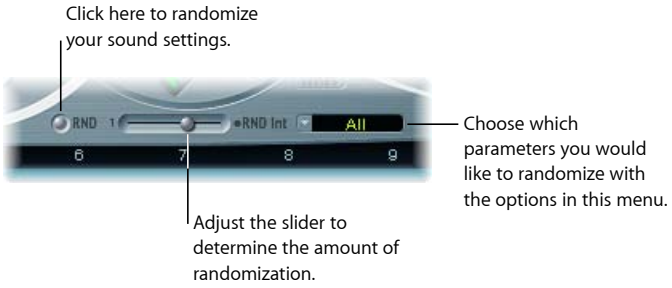
To activate the Phaser effect

- Click the Phaser button.
  - The Intensity parameter determines the depth of the "sweeping" effect—the width of the modulation. Turn this knob to zero to disable the effect.

- The Speed parameter determines the modulation rate.

Creating Random ES2 Sound Variations

The ES2 offers a unique feature that allows you to randomly vary the sound parameters. You can define the amount of random variation and restrict variations to specific sonic elements. The random sound variation feature will inspire and aid you when creating new sounds. It may also occasionally amuse you.



To randomly alter a sound

- Click the RND button below the Filter section.

The process is triggered by a single click and can be repeated as often as you like.

Note: This feature has nothing to do with random real-time modulations. The random feature changes parameter values randomly—each time you click the RND button. Real-time random modulations are achieved with the random LFO waveforms, or through use of the Analog parameter, which randomizes pitch settings.

Tip: It's recommended that you save any sounds resulting from the RND process that you want to keep—as you work. Save the setting with a new name—using the Settings menu of the plug-in window.

Defining the Amount of Randomization in the ES2

You can set the amount of random parameter alteration with the RND Int slider, found to the right of the Random button.

To increase the amount of random variation

- Move the slider further to the right.

The random sound variation feature always alters parameters as they are currently set, not based on the original setting file. Therefore, clicking RND repeatedly will result in a sound that increasingly differs from the original setting.

To create several slight variations of the current setting

- Reload the original setting after each random alteration, saving each with a new name if desired.

Restricting Randomization to ES2 Parameter Groups

Some aspects of your sound may already be ideal for the sound you had in mind. If so, it may not be desirable to alter them. For example, if your sound setting has a nice percussiveness, and you'd like to try a few sonic color variations—while retaining this percussive feel. To avoid the random variation of any attack times, you can restrict the variation to oscillator or filter parameters. You do this by setting the RND Destination to Waves or Filters, thus excluding the envelope parameters from the variation process.

Note: The Master Level, Filter Bypass, and Oscillator On/Off parameters are never randomized. Also, randomizations of the Vector Envelope turn the Solo Point parameter off.

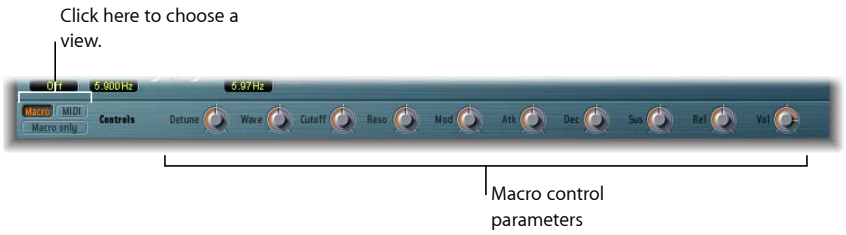
You can restrict random sound variations to the parameter groups listed below:

Parameter group	Comments
All	All parameters, with the exception of those mentioned above, are randomized.
All except router and Pitch	All parameters, with the exception of router parameters and the basic pitch (*semitone* settings of the oscillators), are altered. Oscillator fine-tuning is, however, randomized.
All except Vector Env	All parameters, with the exception of Vector Envelope parameters, are altered. This maintains the rhythmic feel of a given setting.
Waves	Only the oscillator Wave and Digiwave parameters are altered. Other oscillator parameters (tuning, mix, and modulation routings in the *router*) are excluded.
DigiWaves	New Digiwaves are selected for all oscillators. Other oscillator parameters (tuning, mix, and modulation routings in the router) are excluded.
Filters	The following filter parameters are varied: Filter Structure (series or parallel), Filter Blend, Filter Mode, Cutoff Frequency, and Resonance for Filters 1 and 2. The Fatness and Filter FM parameters of Filter 2 are also randomized.
Envs	All parameters of all three envelopes (ENV 1, ENV 2 and ENV 3) are randomized. The Vector Envelope is *excluded*.
LFOs	All parameters of both LFOs are varied.
Router	All router parameters—in all modulation routings—are varied (all intensities, target, via, and source parameters are changed).
FX	All effects parameters are randomized.
Vector Envelope	All Vector Envelope parameters are varied, including the X/Y routing of the Planar Pad.

Parameter group	Comments
Vector Env Mix Pad	The oscillator mix levels (Triangle square icon positions) of the Vector Envelope points are altered. The rhythm and tempo of the modulation (the time parameters of the points) are not changed.
Vector Env XY Pad Options	The Planar Pad square icon positions (the Vector Envelope points) are randomized. The X/Y routing, however, is not changed. The rhythm and tempo of the modulation (the time parameters of the points) are also left unaltered. You can specify a single direction for randomization by choosing either: • Vector Env XY Pad X only • Vector Env XY Pad Y only
Vec Env Times	Only the time parameters of the Vector Envelope points are altered.
Vec Env Structure	The Vector Envelope structure is altered. This includes: All times, the Sustain point, the number of points, and all loop parameters.
Vec Env Shuffle Times	The Vector Envelope shuffle times (within loops) are altered: This includes the Loop Smooth value, if Loop Mode is set to Forward or Backward.

Using Macro Controls and Assigning Controllers in the ES2

The section at the bottom of the ES2 interface provides three views, accessed by clicking the buttons on the left:



- *Macro:* Shows a number of macro parameters that affect groups of other parameters.
- *MIDI:* Allows you to assign MIDI controllers to particular modulation routings (see ES2 Modulation Via Source Reference).
- *Macro only:* Replaces the ES2 interface with a dedicated and much smaller view that is limited to the macro parameters.

Using the ES2 Macro Control Parameters

The macro parameters provide quick access to several linked, related parameters. As you alter any of the macro controls, you will see one, two, or more parameters in the ES2 interface update. For example, adjusting the Detune macro control simultaneously affects the Analog parameter and the coarse and fine oscillator Frequency parameters.

Important: The impact of each macro control is completely dependent on the parameter values of the current setting. In some patches, a number of macro controls may have no effect.

Another bonus of the macro parameters is that they are settings-compatible with ES2-based GarageBand instruments. In other words, you can use the ES2 and some GarageBand synthesizer settings interchangeably.

Assigning Controllers in the ES2

Clicking the MIDI button at the lower left of the interface displays the Controller Assignments. There are six menus, for Ctrl A to Ctrl F. You can use any MIDI controller shown in the menus for these control sources. See Using the ES2 Oscillators.

To assign a controller

- Click a control menu, then choose the controller name/number that you want to use from the list.

To learn a controller assignment via MIDI

1. Click a control menu, then choose the -Learn- item.
2. Move the desired controller on your MIDI keyboard or controller.

Note: If no suitable MIDI message is received within 20 seconds, the selected control will revert to the previous value/assignment.

Nonassignable and 14-Bit Controller Information

Controllers 0 and 32 are reserved for Bank Select messages, controller 1 is used as modulation source in the router, controllers 33 to 63 work as LSB for controllers 1 to 31, controllers 64 to 69 are reserved for pedal messages, controllers 120 to 127 are reserved for channel mode messages.

In the MIDI specification, all controllers from 0 to 31 are known as Most Significant Byte (MSB) controller definitions. Each of these controllers (0 to 31) also contains a Least Significant Byte (LSB) controller definition (32 to 63). Use of this secondary LSB controller in conjunction with the MSB controller allows for a resolution of 14 bits instead of 7 bits. The ES2 recognizes these control change messages—the breath or expression controllers, for example.

To explain:

- 14-bit controllers are pairs of normal Control Change (CC) messages, where the number of the second CC message (the LSB) is 32 higher than the first CC message (the MSB). Examples of valid 14-bit pairs are: CC1/33, CC7/39, and CC10/42.
- 14-bit controllers have a resolution of 16,384 steps, allowing very precise control of plug-in parameters. The first CC message of a 14-bit pair (the MSB) has a coarse resolution of 128 steps. Each of these steps can be divided into a further 128 substeps using the second CC message (the LSB). This results in 128 x 128 = 16,384 steps.
- You don't need to create new, or special, data types to use 14-bit controllers. The finer resolution is achieved by complementing the assigned CC message (the MSB) with its LSB. The CC message assigned in the ES2 can always be used alone if your MIDI controller isn't capable of sending 14-bit messages, thus limiting the resolution to 7-bit = 128 steps.

The 14-bit capability is the reason why CC numbers 33–63 can't be assigned in the Ctrl A–F menus. Using these (LSB) CC numbers would result in changing 1/128th of the parameter range—or put another way, 128 continuous steps out of 16,384.

ES2 Tutorial: Creating Sounds

This tutorial will guide you—from scratch—through the creation of commonly used sounds. The "Using Templates" tutorial will also guide you through the sound creation process, but it starts you off with a number of templates. See ES2 Tutorial: Using Templates to Create Sounds.

To see the settings for these tutorials in the ES2 window, open the Settings menu and choose Tutorial Settings.

ES2 Sound Design from Scratch, Filter Settings, Digiwaves

The *Analog Saw Init* tutorial setting is designed to be used as a starting point when you are programming new sounds from "scratch." When programming entirely new sounds, professional sound designers like to use this type of setting, which has an unfiltered sawtooth wave sound without envelopes, modulations, or any gimmicks. This type of setting is also useful when you are getting to know a new synthesizer. It allows you to access all parameters without having to consider any preset values.

- Start with the filters, the heart of any subtractive synthesizer. Check out the four lowpass filter types—12 dB, 18 dB, 24 dB, and fat (Filter 2)—with different values for Cut (Cutoff Frequency) and Res (Resonance). Define Env 2 as the filter envelope. This modulation wiring is preset in the Router.
- Set Filter Blend to its leftmost position, which will allow you to listen to Filter 1 in isolation. In many circumstances, you'll probably prefer Filter 2, but Filter 1 has its advantages. In addition to the lowpass filter with 12 dB/octave slope (Lo), Filter 2 also offers a highpass, peak, bandpass (BP), and band rejection (BR) mode. Filter 1's lowpass sounds "softer" when compared with Filter 2. It is best-suited to sounds where the filter effect is or should be less audible, such as with Strings and FM sounds. Distorted TB-303-style sounds are more easily achieved with Filter 1.
- This setting is also ideal for checking out the oscillator waveforms. The analog waveforms can be set in the Editor view. To select the Digiwaves, set Osc 1 Wave to Digiwave.

Creating Fat ES2 Sounds with Oscillator Detuning and Unison Mode

"Fat" synthesizer sounds have always been popular and are likely to remain so, given their use in modern trance, techno, R & B, and other styles. The *Analog Saw 3 Osc* setting features three detuned oscillators, and sounds fat as it is. The following will introduce you to some additional tools to fatten the sound even more.

- Check out the 3-oscillator basic sound with different filter and envelope settings.
- Check out the chorus effect at different Intensities and speeds.
- Engage Unison mode and select a higher setting for Analog. Because the sound is polyphonic, each note is doubled. The number of notes that can be played simultaneously will be reduced from 10 to 5. This makes the sound rich and broad. Combining Unison and higher values for Analog spreads the sound across the stereo spectrum.

In many factory settings, the Unison mode is active. This demands a lot of processing power. If your computer isn't fast enough, you can switch off the Unison mode and insert an Ensemble effect in a bus, for use with several plug-ins. This will save processing power. You can also save CPU resources by freezing or bouncing several software instrument tracks.

Creating Detuned Monophonic Sounds and Effects with the ES2

The *Analog Saw Unison* setting is a fat, heavily detuned, unfiltered basic sound. Three sawtooth oscillators are used, but they are further detuned. The combination of Unison and Analog (set to a high value) is essential—but this time monophonic mode is used to stack ten voices. Without further effects, the result is a huge lead sound, much like those used in countless dance and trance productions. With appropriate filter and envelope settings, electro sounds that are ideal for arpeggiation and sequencing can easily be set up.

- Set the Cutoff Frequency of Filter 2 to 0. This activates the preset filter envelope. Feel free to check out different envelope settings.
- Switch Osc 1 to sound one or two octaves lower.
- Increase Drive or Distortion.
- Set Env 2 to be velocity sensitive. This allows for velocity-sensitive filter modulations.
- Insert a delay effect in the instrument channel strip of the ES2 (or a bus target).

Creating Clean Single-Oscillator Bass Sounds with the ES2

Not every sound needs to consist of several oscillators. There are numerous simple, effective, sounds that make use of a single oscillator. This is especially true of synthesizer bass sounds, which can be created quickly and easily with the *Analog Bass Clean* setting.

The basic sound is a rectangular wave, transposed down by one octave. The sound is filtered by Filter 2. What's special about this sound is its combination of Legato and Glide (portamento). When you play staccato, no glide effect will occur. When you play legato, the pitch will smoothly glide from one note to another. All keys must be released before you strike a new key, in order to retrigger the envelopes.

- Check out different filter and envelope settings.
- Replace the rectangular wave with a sawtooth.
- Vary the Glide settings.

It's best to make your edits while a bass line is playing. Create or play a monophonic bass line, with most notes played staccato, but some legato. This can provide some interesting results with very long Glide values.

Creating Distorted Analog Basses with the ES2

In the *Analog Bass Distorted* setting, Filter 1 is engaged, with high settings for Drive and Distortion. This filter is better suited to the creation of distorted analog sounds than Filter 2.

- Check out Filter 2 by setting Filter Blend to its rightmost position. Notice that Filter 1 works better with distorted sounds.

- To control the filter modulation, move the green sliders of the first modulation channel in the router. This controls the modulation intensity.

Using FM Intensity and Frequency to Create ES2 Sounds

The *FM Start* setting is great for familiarizing yourself with linear frequency modulation (FM) synthesis. You'll hear an unmodulated sine sound, generated by Oscillator 1. Oscillator 2 is switched on and set to produce a sine oscillation as well, but its level is set to 0: Drag the small square in the uppermost corner of the Triangle to change the settings.

In the ES2, Oscillator 1 is always the carrier, and Oscillator 2 the modulator. In other words, Oscillator 2 modulates Oscillator 1.

- Adjust the intensity of the frequency modulation by slowly moving the wave selector from Sine to FM. You will hear a typical FM spectrum, with the carrier and modulator set to the same frequency.
- Alter the modulator frequency (Oscillator 2) by adjusting Fine Tune from 0 c to 50 c. You'll hear a very slow frequency modulation, which can be compared to the effect of an LFO. The frequency modulation, however, takes place in the audio spectrum. It is adjusted in semitone steps by the frequency selector. Check out the entire range from –36 s to +36 s for Oscillator 2. You'll hear a broad spectrum of FM sounds. Some settings will remind you of classic FM synthesizer sounds.
- Select other waveforms for Oscillator 2. Sine is the classic, standard FM waveform, but other waveforms lead to interesting results as well, especially the Digiwaves.
- You will achieve further interesting results by altering the carrier (Oscillator 1) frequency. Check out the entire range, from –36 s to +36 s here, as well. The odd intervals are especially fascinating. Note that the basic pitch changes when you do this.

Controlling ES2 FM Intensity with an Envelope and FM Scaling

In the *FM Envelope* setting, you can control the FM intensity with an envelope, generated by Envelope 2. The modulation target is the range that falls between Sine and FM in the Oscillator wave selector. The first Router channel is used for this modulation routing. You can control a wider range by using additional modulation routings, which have been pre-prepared for you. All you need to do is set their values. As these modulations work without velocity sensitivity, you can set them in the Editor view by moving both the lower and upper fader halves to their topmost positions.

- Set the second modulation channel to 1.0. You'll hear how the modulation now "wanders" through a broader sound range.
- Set modulation channels 3 and 4 to a value of 1.0 as well, and listen to the increase in the sound range.

- After these drastic augmentations to the modulation range, the sound will become uneven across the keyboard. In the lower and middle ranges it sounds nice, but in the upper key range the FM intensity appears to be too severe. You can compensate for this effect by modulating the Osc 1 Wave target by keyboard position (kybd) in modulation routings 5 and 6. This results in a keyboard scaling of the FM intensity.
- As the sound range is so vast (due to the 4 modulations), two modulation routings are required to compensate for this. Set the lower slider halves to their lowest positions. Good keyboard scaling is essential for any FM sound.

Using FM Drive and Filter FM to Change the Color of ES2 Sounds

The *FM Drive* setting illustrates how dramatically the character of FM sounds can be altered when you apply Drive and Filter FM. The results are reminiscent of the feedback circuits of classic FM synthesizers.

- Check out different Drive and Filter FM settings.
- Lower the Cutoff Frequency of Filter 2 to 0. Envelope 2 modulates Filter 2. This modulation routing is already present in the setting.

Creating FM Sounds with Digiwaves in the ES2

In the *FM Digiwave* setting, a Digiwave is used as an FM modulator. This results in bell-like spectra from only two operators. With traditional FM synthesis, this type of timbre could normally be produced only with a larger number of sine oscillators.

To create a fatter, undulating, and atmospheric quality to the sound, the polyphonic Unison mode has been engaged. Filter and amplitude envelopes have been preset to shape the sound.

- Check out the variety of Digiwaves as FM modulation sources.
- Check out different Analog parameter values.

Creating FM Sounds with Wavetables in the ES2

You can program the most vivid FM sounds when the modulation source morphs between different Digiwaves. The morphing in the *FM Digiwave* setting is controlled by LFO 2. The tempo of LFO 2, and therefore the morph, depends on the host application tempo—here, 2 bars.

- Set LFO 2 to different waveforms. Lag S/H (smooth random), in particular, should be fun.
- Check out different FM intensities and oscillator frequencies.
- Alter the modulation intensity of the first modulation channel (LFO2 modulates Osc2 Wave) and the LFO 2 rate.

Creating Distorted ES2 FM Sounds with Monophonic Unison

The *FM Megafat* setting is well-suited for distorted basses and guitar-like sounds. This sound gets rather "rude" in the upper key range. This cannot be compensated for with key scaling, but not every sound has to be "nice" across the entire keyboard range!

- Check out extreme detunings by adjusting the Analog parameter.
- Check out the Flanger with this sound.
- Engage the filter envelope by lowering the Cutoff Frequency of Filter 2 down to 0.
- Add some Glide to lead sounds.
- As always when it comes to FM, you can dramatically alter the sound by varying the frequencies of the oscillators. Make sure you check out the odd intervals, as well.

Creating FM Sounds with Unusual Spectra in the ES2

If you're unconcerned with the pitch of your sound, you can get the weirdest spectra out of odd frequency ratios—oscillator intervals.

The *FM Out of Tune* setting offers a bell-like sound, reminiscent of a ring modulator. It was achieved through a setting of 30 s 0 c, with the modulator set to a value of 0 s 0 c. Sounds like this were commonly used in the electronic music of the eighties and have had a resurgence in popularity in ambient and trance music styles.

You can further develop the sound by applying filtering, envelope modulations, and effects. There is, however, one small problem—the sound is out of tune.

- Use Oscillator 3 as a reference for the tuning of the FM sound by dragging the square icon in the Triangle.
- You'll notice that the sound is 5 semitones too high (or 7 semitones too low, conversely).
- Transpose both Oscillators 1 and 2 five semitones (500 ct) lower. Transposing them upward is not practical, as you'd need to select 37 s 0 c for Oscillator 1, which has a maximum value of 36 s 0 c.
- It's important to maintain the frequency ratio (interval) between Oscillators 1 and 2. This means that Oscillator 1 will sound at 25 s O c and Oscillator 2 at –5 s 0 c.

Setting Pulse Width Modulations with Oscillator 2 in the ES2

Pulse width modulation (PWM) is one of the most essential features of any analog synthesizer.

- Choose the *PWM Start* setting, and move the Wave control slowly back and forth between the rectangular and the pulse wave symbols. Both are green. What you will hear is a manual pulse width modulation.
- Choose the *PWM Slow* setting. Here, LFO 1 controls the pulse width modulation source, not your manual movements. The result should be quite similar.

- Raise the LFO 1 rate from its preset value of 0.230 to 4.400. The result is a classic, fast PWM.
- In this and the next step, the PWM will be set so that it sounds slower in the lower keyboard range and faster in the upper range. This is desirable for many sounds, such as synthetic strings. First, reduce the LFO 1 Rate to 3,800.
- Change the modulation intensity of the second router channel (target = LFO1 Rate, Source = Kybd) to 0.46. This will alter the scaling of the PWM, making it sound faster in the treble range. You can also hear this type of effect in the *PWM Scaled* setting.

  Tip: Avoid Drive and Distortion with PWM sounds.

Creating ES2 String Sounds with Pulse Width Modulation

To make the sound fatter, add Oscillator 3, which can also be pulse width modulated. In fact, even the first oscillator can deliver PWM. In the *PWM 2 Osc* setting, both oscillators are detuned quite significantly. Develop your own personalized PWM string sound, using this setting as your base.

- Adjust the Chorus intensity. You'll probably choose higher values, which make the sound rather broad.
- Program Envelope 3 according to your taste. You should, at the very least, raise the attack and release times. Define it to react to velocity, if you prefer. If you want to use the sound for something other than a simple pad, a shorter Decay Time and a lower Sustain Level of about 80 to 90% may be more appropriate.
- Reduce the Cutoff Frequency and Resonance of Filter 1 to make the sound softer.
- Save the new setting.
- Compare the result with the original PWM 2 Osc setting. You'll hear that the sound has undergone a remarkable evolution.
- Also compare it to *PWM Soft Strings*, which was created as described above. You'll probably notice a few similarities.

Creating Sounds with Ring Modulation in the ES2

A ring modulator takes its two input signals and outputs their sum and difference frequencies.

In the ES2, Oscillator 2 outputs a ring modulation, which is fed with a square wave of Oscillator 2 and the wave of Oscillator 1, when Ring is set as Oscillator 2's waveform.

Odd intervals (frequency ratios) between the oscillators result in bell-like spectra, much like those heard in the *Ringmod Start* setting.

The third oscillator can be used as a tuning reference, to maintain a kind of basic tuning. On occasion, you may find that it's nice to leave the sound out of tune—for use as a source of overtones and harmonics for another basic wave, supplied by Oscillator 3.

Try to program an atmospheric bell sound. Use your imagination, but here are some hints that may help:

- Experiment with the various frequency ratios of Oscillators 1 and 2. You may want to use the 29 s 0 c/21 s 0 c ratio, which doesn't sound out of tune at all. Ring modulation is not only useful for bell-like sounds, it's also good for a great variety of spectra that tend to sound weird at lower frequency settings. Also try alterations to the fine-tuning of the Oscillators.
- Check out an Intensity of 50% and a Rate, set to around 2/3 of the maximum value, for the Chorus effect.
- Set the Attack and Release Times of Envelope 3 to taste.
- Check out Drive and Filter FM, if you like your sounds a little "out of control."
- The rest is up to you.

Creating Sounds with Oscillator Synchronization in the ES2

If you select the synced square and sawtooth waveforms for Oscillators 2 and 3, they will be synchronized with Oscillator 1. In the *Sync Start* setting, only Oscillator 2 is audible, and Oscillator 3 is switched off.

Typical sync sounds feature dynamic frequency sweeps over wide frequency ranges. These frequency modulations (the sweeps) can be applied in various ways.

- Try the pre-programmed pitch modulation, assigned to the modulation wheel first.
- In the second router channel, an envelope pitch modulation has been preprogrammed (target = Pitch 2, Source = Env 1). Setting the minimum value to 1.0 results in a typical sync envelope. Also check out shorter Decay Times for Envelope 1.
- To avoid a sterile, lifeless sound (after the decay phase of the envelope), you may also want to modulate the oscillator frequency with an LFO. Use the third router channel, and set the minimum modulation applied by LFO 1 to about 0.50.
- Substitute the synchronized square wave with the synced sawtooth wave, and see if you like the results.

Note: Pulse width modulation is also available via the synchronized square wave of Oscillators 2 and 3. A modulation of the wave parameters of these two oscillators results in a PWM when the synced square wave is selected.

Getting Started with Vector Synthesis in the ES2

This tutorial section provides some hints for Vector Envelope programming. In the *Vector Start* setting, the "mix" of the oscillators is controlled by the Vector Envelope. Each oscillator has been set to a different waveform.

- Switch from the Router view to Vector view.

- In its basic (default) setting, the Vector Envelope has 3 envelope points. Point 1 is the start point, point 2 the Sustain point, and point 3 is the target in the release phase. By clicking the points, you can see that the mix is always set to 100% for Oscillator 1, in the Triangle.
- Click point 2, and drag the square icon in the Triangle to Oscillator 2. You'll hear a square wave, instead of Oscillator 1's sawtooth.
- Engage the Vector Envelope by switching the Solo Point parameter off. When it is switched on, you hear only the selected point, with no dynamic modulation. When Solo Point is switched off, you'll hear the sound moving from saw to square, with every triggered note.
- Alter the preset time of 498 ms between points 1 and 2.
- While holding Shift, click between points 1 and 2. This will create a new point 2, and the point formerly known as point 2 will become point 3. The total time span between point 1 and point 3 is divided into the times between points 1 and 2, and 2 and 3. The division takes place at the click location. If you click at the exact midpoint, the new time spans are equal.
- Click the newly created point 2, and then drag its corresponding square icon in the Triangle to Oscillator 2.
- Click point 3, and drag its corresponding square icon in the Triangle to Oscillator 3. Listen to the three oscillators morphing from sawtooth to square to a triangular wave at the final Sustain point.
- Click point 4 (the end point) and drag its corresponding square icon in the Triangle to Oscillator 1, if it's not already there. Listen to how the sound returns to Oscillator 1's sawtooth wave, following the release of the key.

Using the Planar Pad for Vector Synthesis in the ES2

The *Vector Envelope* setting starts where the *Vector Start* setting left off. You have a simple Vector Envelope consisting of 4 points, which is set to modulate the oscillator mix (the Triangle).

In this example, the Vector Envelope will be used to control two additional parameters—the Cutoff Frequency of Filter 2 and Panorama. These are preset as the X and Y targets in the Planar Pad. Both have a value of 0.50.

- Switch on Solo Point, to more easily listen to the settings for the single points.
- Click point 1. You will hear only Oscillator 1's sawtooth.
- Drag the square icon in the Planar Pad to the far left, which results in a low cutoff frequency for Oscillator 2.
- Click Point 2. You will hear only Oscillator 2's rectangular wave.

- Drag the square icon in the Planar Pad all the way down, which results in the rightmost panorama position.
- Click Point 3. You will hear only Oscillator 3's triangular wave.
- Drag the square icon in the Planar Pad all the way up, which results in the leftmost panorama position.
- Switch on Solo Point. The sound begins with a strongly filtered sawtooth wave and turns into an unfiltered square wave. It initially sounds from the right, and then it moves to the left while morphing into a triangular wave. After you release the key, the saw sound will be heard.

Using Vector Synthesis Loops in the ES2

The basic sound of the *Vector Loop* setting—without the Vector Envelope—consists of three elements:

- Oscillator 1 delivers a metallic FM spectrum, modulated by Oscillator 2's wavetable.
- Oscillator 2 outputs cross-faded Digiwaves (a wavetable), modulated by LFO 2.
- Oscillator 3 plays a PWM sound at the well-balanced, and keyboard-scaled, speed of LFO 1.

Unison and Analog make the sound fat and wide.

These heterogeneous sound colors will be used as sound sources for the vector loop.

A slow, forward loop is preset. It moves from Oscillator 3 (PWM sound, point 1) to Oscillator 1 (FM sound, point 2), then to Oscillator 3 again (PWM, point 3), then to Oscillator 2 (wavetable, point 4), and finally it returns to Oscillator 3 (PWM, point 5). Points 1 and 5 are identical, which prevents any transition from point 5 to point 1 in the forward loop. This transition could be smoothed out with Loop Smooth, but this would make the rhythmic design more difficult to program.

The distances between the points of the Vector Envelope have been set to be rhythmically exact. Given that Loop Rate has been engaged, the time values are not displayed in ms, but as percentages. There are four time values (each at 25%), which is a good basis for the transformation into note values.

- Switch off the Vector Envelope by setting Solo Point to on. This allows you to audition the individual points in isolation.
- Take the opportunity to alter the square icon positions in the Planar Pad according to your taste. The X/Y axes of the Planar Pad control the cutoff frequency of Filter 2, and the panorama position. Adjustments to these make the sound more vivid.
- Activate the Vector Envelope by setting Solo Point to off. Check the result, and fine-tune the square icon positions in the Planar Pad.

- Alter the Loop Rate from the preset value of 0.09 up to 2.00. You will hear a periodic modulation, much like that of an LFO. At this point, the modulation is not synchronized with the project tempo. To synchronize the loop speed with the project tempo, move the Rate to the far left, and set a note or bar value.
- You can create faster rhythmic note values by clicking between two points and setting the new time values—which result from the division that occurs—to a value of 12.5%, for example.

Creating ES2 Bass Drum Sounds with a Self-Oscillating Filter and the Vector Envelope

Electronic kick drum sounds are often created with modulated, self-oscillating filters. This approach can also be taken with the ES2, particularly when the Vector Envelope is used for filter modulation. An advantage of the Vector Envelope, in comparison with conventional ADSR envelopes, is its ability to define and provide two independent decay phases. The distortion effect applies the right amount of drive without sacrificing the original sonic character of the drum sound.

Note: To make the *Vector Kick* setting really punchy, you must activate Flt Reset, because all oscillators are switched off in this setting, and the filter needs a little time to start oscillating. At the start of each note, Flt Reset sends a very short impulse to the filter—making it oscillate from the outset.

By tweaking the *Vector Kick* setting you'll be able to create any dance-floor kick drum sound your heart desires. These are the parameters that allow for the most efficient and significant variations:

- Filter 2 slopes: 12 dB, 18 dB, 24 dB
- Distortion: Intensity and Soft or Hard
- Envelope 3's Decay Time: (D)
- Vector Envelope Time 1 > 2: preset to 9.0 ms
- Vector Envelope Time 2 > 3: preset to 303 ms
- Vector Time Scaling

Creating Percussive ES2 Synthesizer and Bass Sounds with Two Filter Decay Phases

As with the Vector Kick setting, the *Vector Perc Synth* setting uses the Vector Envelope to control the filter cutoff frequency, with two independently adjustable decay phases. This would not be possible with a conventional ADSR envelope generator. Try creating further percussive synthesizers and basses by varying these parameters:

- Vector Envelope Time 1 > 2 (= Decay 1)
- Vector Envelope Time 2 > 3 (= Decay 2)

- Vector Time Scaling
- Square icons in the Planar Pad for points 1, 2, and 3 (= Cutoff Frequency)
- Waveforms (choosing other waveforms)

ES2 Tutorial: Using Templates to Create Sounds

Welcome to a brief programming tour of the ES2!

While working on the factory preset programming for the ES2, a number of testers, sound programmers, and other people involved in the project indicated that it would be nice to start their programming work from templates, rather than entirely from scratch. This feedback resulted in a number of tutorial templates that were added to the Settings menu of the ES2 window (open the Settings menu and choose Tutorial Settings to see these templates).

Needless to say, creating templates that cover all sound genres is an impossible mission. As you spend time familiarizing yourself with the ES2's architecture, you'll start to understand why.

Nevertheless, this programming tour of the ES2 is included as a part of the toolbox to help you learn the ES2's architecture through experimentation. You'll find that this approach is fun. You'll also discover, as you work through a number of simple operations, that results come quickly when you start to create your personal sound library.

As you become more familiar with the ES2 and its myriad functions and parameters, you can create your own templates to use as starting points for designing new sounds.

Using the ES2 Slapped StratENV Setting

The target of this setting is the sound of a Stratocaster, with the switch between bridge and middle pickup in the middle position—in phase. It attempts to emulate the noisy twang typical of this sound's characteristics.

This might be a useful template for emulations of fretted instruments, harpsichords, clavinets, and so on.

Take a look at its architecture:

Osc 1 and Osc 3 provide the basic wave combination within the Digiwave field. Changing the Digiwaves of both in combination delivers a huge number of basic variations—some also work pretty well for electric piano-type keyboard sounds.

Osc 2 adds harmonics with its synced waveform, so you should only vary its pitch or sync waveform. There are a couple of values that can be changed here, which will give you a much stronger, more balanced signal.

An old trick, which delivers a punchy attack, was used—to create an effect that the use of a naked wave wouldn't deliver, even with the best and fastest filters available: You use an envelope (in this case, Env 1) for a quick "push" of a wavetable's window—or all wavetables together, where it makes sense.

Set up Envelope 1's decay time for this short push by moving the wave selectors for all oscillators. (Although it makes no sense to do this on the synced sawtooth oscillator, Osc 2, use the envelope trick regardless.)

This allows you to vary the punchiness of the content between:

- Envelope 1's contribution to the overall attack noise and changing decay length—a short decay results in a peak, a long decay results in a growl, as Envelope 1 reads a couple of waves from the wavetable.
- Modulation destination—you can always assign this to each of the oscillators separately.
- Start point—you vary the wave window start with minimum and maximum control of EG1/Osc.waves modulation: negative values for a start wave before the selected wave, positive values for a start wave from a position behind the selected wave that rolls the table back.

Feel free to experiment with this wavetable-driving trick. The growl effect works well for brass sounds, and some organs absolutely shine with a little click, courtesy of a wavetable push.

Envelope 2, which controls the filter, provides a slight attack when used for "slapped" characteristics. Setting it to the fastest value eliminates the wah-like attack, while retaining the punch.

For playing purposes, you'll find that LFO 2 is used as a real-time source for vibrato. It is assigned to the mod wheel and pressure.

Don't concern yourself too much with the different wheel and pressure settings. Feel free to change them.

Velocity is set up to be very responsive, because many synthesizer players don't strike keys like a piano player would with a weighted-action "punch." Therefore, you should play this patch softly, or you may find that the slap tends to sweep a little. Alternatively, you can adjust the sensitivity of the filter modulation's velocity value to match your personal touch.

Feel free to increase the Voices to maximum—six strings should be enough for a guitar, but for held or sustained notes, a few extra voices may come in handy.

Using the ES2 Wheelrocker Setting

This quite ordinary organ patch doesn't hold any deep, high-end sound design secrets—it is just a combination of three oscillators with mixed wave levels. You'll probably find a different combination that more closely matches your vision of what an organ sounds like. Check out the Digiwaves.

Focus your attention on the mod wheel's response—hold a chord, and bring the wheel in by moving it slowly upward until you reach the top (maximum).

The intention behind the programming of this mod wheel modulation is to simulate an accelerating Leslie rotor speaker.

The modulation routings do the following:

- Modulation routing 1 assigns envelope 2 to Filter 1—the only one used for this patch—and produces a little organ key click with the envelope. The filter is opened slightly (with Keyboard as via) when you play in the higher keyboard range, with the maximum value.
- Modulation routings 2 and 3 bring in LFO 1 vibrato, and both oscillators are modulated out of phase.
- Modulation routing 4 does not need to be adjusted, but you are free to do so. It has been set up to use ENV1 to "push" the wavetable. Adjust ENV1 Decay to make the sound more pipe organ-like. Adjust ENV1 Attack to sweep through the wavetable.
- Modulation routing 5 reduces the overall volume according to personal taste, but the organ's level shouldn't increase too drastically when all modulations are moved to their respective maximums.
- Modulation routings 6 and 7 detune Oscillators 2 and 3 against each other, within symmetrical values—to avoid the sound getting out of tune, overall. Again, both work out of phase with modulation routings 2 and 3; Oscillator 1 remains at a stable pitch.
- Modulation routing 8 brings in LFO 1 as a modulator for panorama movement—this patch changes from mono to stereo. If you prefer a full stereo sound with a slowly rotating Leslie in its idle position, just set an amount equal to the desired minimum value, thereby achieving a permanent, slow rotation. Another modification you may wish to try is a higher value, resulting in more extreme channel separation.
- Modulation routing 9 speeds up LFO 2's modulation frequency.
- Modulation routing 10: A little cutoff was added to Filter 1, increasing the intensity of the big twirl.

Feel free to find your own values. While doing so, keep in mind the fact that there are two modulation pairs that should only be changed symmetrically—modulation routings 2 and 3 work as a pair, as do modulation routings 6 and 7. So, if you change Pitch 2's maximum to a lower minus value, remember to set Pitch 3's maximum value to the same positive amount—the same rule applies for modulation routing pair 6 and 7.

You can also bring in LFO 2 to increase the pitch diffusion against LFO 1's pitch and pan movements. Just exchange it for LFO 1 on modulation 2 and 3. Note that there will be no modulation source for the Leslie acceleration, so you'll need to use it in a static way, just fading it in. Alternatively, you'll need to sacrifice one of the other modulations in favor of a second twirl.

For another stereo modification of the static sound, you can use the patch in Unison mode with a slight detune—make sure to adjust the analog parameter for this.

Using the ES2 Crescendo Brass Setting

The oscillators are used for the following tasks:

- Oscillator 1 provides the basic brass wave—sawtooth.
- Oscillator 2 provides a not particularly "brassy" pulse wave, which brings in the ensemble. It is pulse-width modulated by LFO 1 (modulation routing 4).

Note: The following critical point should be taken into account for any modulations. There are four parameters that behave in an entirely different fashion when any one of them is changed. Therefore, all four must be changed when making adjustments:

- You may adjust the initial pulse width of Oscillator 2's wave parameter. A "fat" position, close to the ideal square wave, has been chosen for this patch—in order to program a full, voluminous synth-brass sound.
- Modulation routing 4 adjusts the modulation intensity—how far the range differs from fat to narrow when being pulse-width modulated. Set with the Minimum parameter.
- The rate of LFO1 directly controls the speed of the movement of the pulse width modulation. For this patch, both LFOs are used, to achieve a stronger diffusion effect at different modulation speeds.

  Tip: You should use LFO1 for all permanent, automatic modulations, because you are able to delay its impact with its EG parameter. You can use LFO 2 for all real-time modulations, which you intend to access via ModWheel, pressure, or other controls while playing.

- A keyboard assignment was set up as the source for modulation routing 4. This is because all pitch, or pulse-width, modulations tend to cause a stronger detuning in the lower ranges, while the middle and upper key zones feature the desired diffusion effect. When using this parameter, you should initially adjust the lower ranges until an acceptable amount of detuning (resulting from the modulation) is reached. When set, check whether or not the modulations in the upper zones work to your satisfaction. Adjust the relationship between intensity (Max) and scaling (Min) values.

Oscillator 3 generates a Digiwave, which is "brassy" enough, within the overall wave mix. As an alternative to the Digiwave, another modulated pulse wave could be used to support the ensemble, or another sawtooth wave—to achieve a "fatter" sound, when detuning it with Oscillator 1's sawtooth wave.

The primary aim, however, is to have a little bit of "growl," achieved through a short wavetable push, as described in Using the ES2 Slapped StratENV Setting. This configuration is set up in modulation routing 3 (Oscillator 3 Wave moved by Envelope 1's Decay).

Other controls have a variety of functions:

- Envelope 1 also effects the pitch of Oscillator 2 against Oscillator 3. This results in both pitches clashing with each other, and also with the stable pitch of Oscillator 1 in the attack phase of the sound.
- The filter envelope's design closes with a short stab in the attack phase, then reopens for a slower crescendo phase.
- A further real-time crescendo has been assigned to the mod wheel, which also brings in an overall pitch modulation, controlled by LFO 2.
- In addition to all of this, a "contrary" real-time modulation by pressure—which closes the filters—has been programmed. This allows you to play with an additional decrescendo, remotely controlled by touch. Try to get a feel for the patch's response. You'll find that it offers quite a few controls for expression—velocity, pressure after note-on, and pressure in advance. Listen to what happens when you press with the left hand before hitting a new chord with the right hand and allowing the swell to come in.

Using the ES2 MW-Pad-Creator Setting

This is an attempt to create a patch that is able to automatically generate new patches.

Again, Oscillator 2 is used for a pulse width modulation—which creates a strong ensemble component (for more information, see Using the ES2 Crescendo Brass Setting.

Oscillators 1 and 3 are set to an initial start wave combination within their respective Digiwave tables. You can modify these, if you wish, and start with a different combination of Digiwaves from the outset.

Modulation 3 "drives" the wavetables of all three oscillators, via the mod wheel. Stated simply, you can simultaneously scroll through the Oscillator 1 and Oscillator 3 wavetables, and change Oscillator 2's pulse width—by moving the mod wheel.

Try a careful, very slow movement of the mod wheel, and you'll hear drastic changes within the wave configuration. Each incremental position of the wheel offers a different digital pad sound. Avoid rapid movements, or this will sound like an AM radio.

Another potential modification procedure is hidden in the modulation intensity of the Oscillator 1, 2, and 3 wave parameters. As mentioned for the Slapped Strat setting, the value of this intensity parameter assigns both the step width, and direction, through the wavetables. You can try modifications to the amount, using positive or negative values.

An interesting side-effect of FM assignment to Filter 2 (modulation routing 4—Lowpass Filter FM) occurs when the mod wheel is moved to higher positions: the frequency modulation of the filter is increased, causing all cyclical beats (vibrating pitches, detunes, pulse width) to be emphasized. This also adds a rough, "hissing" quality to the overall sound character.

FM offers vast scope for experimentation, and you can decide between:

- An initial FM, using Filter 2's FM parameter, which you can redraw (set a negative modulation amount for modulation routing 4's maximum) by moving the mod wheel to its top position.
- Permanent FM (and another modulation setup, saved for a different assignment). You can also switch off FM, if you consider its effect too dirty sounding.

Real-time control is via pressure for a vibrato (modulation routing 10), and also for a slight opening of the Cutoff to emphasize the modulation (modulation routing 9).

Using the ES2 Wheelsyncer Setting

Never obsolete—and undergoing a renaissance in new popular electronic music—are sync sounds.

For a description of the technical aspects of forcing an Oscillator to sync, see Synchronizing the ES2 Oscillators. Here's the practical side of the playground.

Wheelsyncer is a single-oscillator lead sound; all others are switched off.

Although Oscillator 2 is the only one actively making any sound, it is directly dependent on Oscillator 1.

If you change Oscillator 1's pitch or tuning, the overall pitch of the sound will go out of tune or will be transposed.

The pitch of Oscillator 2 provides the tone-color (or the harmonics) for the sync sound. Pitch changes are controlled by modulation routing 7—Oscillator 2 pitch is assigned to the mod wheel.

If you move the wheel, you can scroll through the spectrum of harmonics that have been programmed—for real-time changes. Any modification here starts with the pitch of Oscillator 2 itself, which is set to 3 semitones below the overall pitch. Feel free to start with a different pitch for Oscillator 2; it won't affect the tuning of the sound.

The next modification may be modulation routing 7's intensity (or the interval). The maximum value has been chosen—if this is too extreme for your needs, feel free to reduce it.

Another modification lies in the tone color of the lead sound itself. Oscillator 1 is switched off, because the patch is OK as it is. If you switch it on, all Oscillator 1 waveforms—including Digiwaves, standard waveforms, or a sine wave (which can be further modulated by FM)—are available for use.

All real-time controls are via the mod wheel, which is used for opening the filter on modulation routing 6, a panning movement on modulation routing 8, and acceleration of panning movement on modulation routing 9. If you have deeper modulation ambitions, a similar setup is used for a Leslie speaker simulation in the Wheelrocker setting (see Using the ES2 Wheelrocker Setting.

EFM1

6

The 16-voice EFM1 is a simple, but powerful, frequency modulation synthesizer. It can produce the rich bell and digital sounds that frequency modulation (FM) synthesis has become synonymous with.

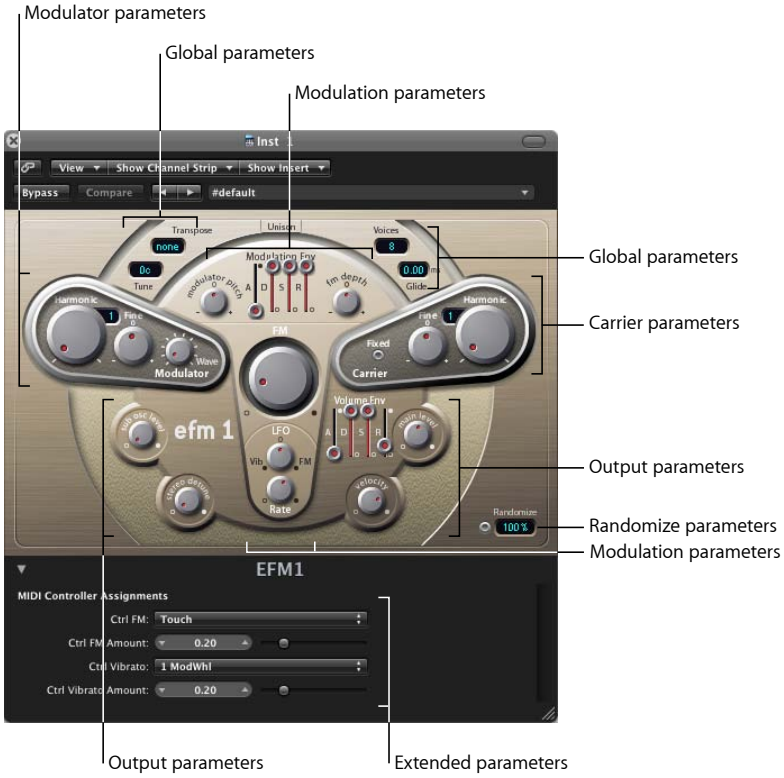
If you're totally new to synthesizers, it might be best to start off with Synthesizer Basics which will introduce you to the terminology and give you a great overview of different synthesis systems and how they work.

This chapter covers the following:

- Getting to Know the EFM1 Interface (p. 132)
- Working with the EFM1 Modulator and Carrier Parameters (p. 133)
- Working with the EFM1 Modulation Parameters (p. 136)
- Adjusting Global EFM1 Parameters (p. 137)
- Setting EFM1 Output Parameters (p. 138)
- Creating EFM1 Random Sound Variations (p. 139)
- Assigning EFM1 MIDI Controllers (p. 139)

Getting to Know the EFM1 Interface

Before you take a look at the individual parameters of the EFM1, this section will familiarize you with the various elements that constitute the EFM1 graphical interface.



The EFM1 is divided into several areas.

- *Global parameters:* The top section contains parameters that set the overall tuning of the EFM1. Further controls enable you to set the Glide (portamento) time, limit the number of voices, and to thicken the sound with Unison. See Adjusting Global EFM1 Parameters.
- *Modulator and Carrier parameters:* The FM engine consists of the Modulator and Carrier parameters (raised, darker sections), and the FM Intensity knob (in the center). These are the key controls for setting the basic tone of the EFM1. See Working with the EFM1 Modulator and Carrier Parameters.
- *Modulation parameters:* The modulation envelope and LFO shown at the top and bottom of the mushroom-shaped area in the center, respectively, are used to animate the sound. See Working with the EFM1 Modulation Parameters.

- *Output parameters:* The bottom section houses the Output section, which features the Sub Osc Level and Stereo Detune knobs, that can be used to thicken the sound. The volume envelope, Main Level, and Velocity controls are used to set the EFM1 level. See Setting EFM1 Output Parameters.
- *Randomize parameters:* The Randomize field and button are shown to the lower right. They are used to create random variations of the current settings, resulting in new sounds. See Creating EFM1 Random Sound Variations.
- *Extended parameters:* Accessed by clicking the disclosure triangle at the lower left of the interface, these parameters allow you to assign MIDI controllers to the FM Depth and Vibrato parameters. See Assigning EFM1 MIDI Controllers.

Working with the EFM1 Modulator and Carrier Parameters

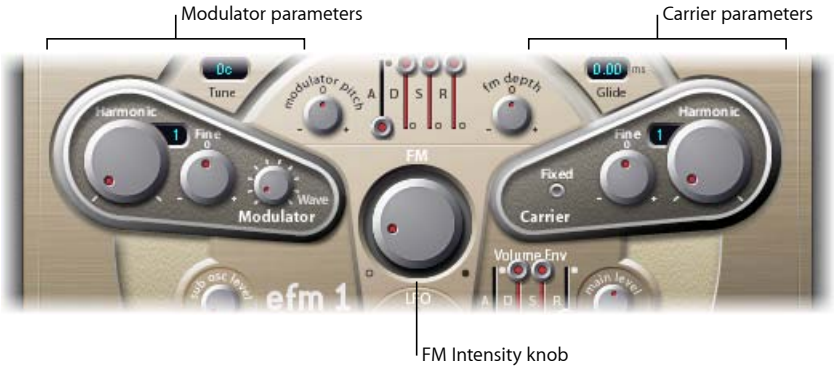
In FM synthesis, the basic sound is generated by setting different tuning ratios between the Modulator and Carrier oscillators, and by altering the FM intensity. The tuning ratio determines the basic overtone structure, and FM intensity controls the level of these overtones.

At the core of the EFM1 synthesis system, you'll find a multi-wave *modulator* oscillator and a sine wave *carrier* oscillator. The basic sine wave (of the carrier oscillator) is a pure, characterless tone.

To make things more sonically interesting, the modulator oscillator is used to modulate the frequency of the carrier oscillator. This modulation occurs in the audio range (you can actually hear it), and results in a number of new harmonics becoming audible.

The pure sine wave (of the carrier oscillator) is combined with the newly generated harmonics, making the sound much more interesting.

Changes to the ratio of the two oscillators is achieved by adjusting the Harmonic parameters, found in both the Modulator and Carrier sections. Additional tuning control is provided by the Fine (Tune) parameters, discussed below.



- *Harmonic knobs:* Set the tuning ratio between the Modulator (left) and Carrier (right) oscillators. See Setting the EFM1 Tuning Ratio.
- *Fine (tune) knobs:* Adjust the tuning between two adjacent harmonics (as determined by the Harmonic knobs). The range of this control is the center (0) position, Fine tune does not have an effect. Click the "0" to center the Fine tune knob. Dependent on the amount of detuning, you will hear either:
  - A subtle "beating" of the timbre-if lower detuning amounts are used.
  - New harmonic and inharmonic overtones-if high detuning amounts are used.
- *FM (Intensity) knob:* Sets the amount of carrier oscillator frequency modulation by the modulator oscillator. As you adjust the FM knob, the intensity (and number) of newly generated overtones (harmonics) increases-making the sound brighter.

  Note: Although the technology behind it is very different, you could compare the FM (Intensity) parameter with the Filter Cutoff parameter of an analog synthesizer.
- *Wave knob (Modulator):* Chooses a different waveform for the modulator oscillator. See Choosing a Different EFM1 Modulator Waveform.
- *Fixed Carrier button (Carrier):* Enable to disconnect the carrier frequency from keyboard, pitch bend, and LFO modulations. This allows you to produce a carrier tone that is free of these modulation sources.

Setting the EFM1 Tuning Ratio

The carrier frequency is determined by the played key, and the modulator frequency is typically a multiple of the carrier frequency.

You can tune the modulator and carrier to any of the first 32 harmonics. The tuning relationship (or ratio) between the two significantly changes the base sound of the EFM1, and is best set by ear.

The Harmonic knobs are used to set the tuning ratio between the Modulator (left) and Carrier (right) oscillators.

As a rule of thumb: even tuning ratios between the carrier and modulator tend to sound more harmonic or musical, while odd ratios produce more inharmonic overtones—which are great for bell and metallic sounds.

In this respect, you can view the tuning ratio as being somewhat like the waveform selector of an analog synthesizer.

Note: The Harmonic and Fine tune knobs only affect the tuning relationship between the Carrier and Modulator oscillators. These should not be confused with the global Tune and Fine Tune parameters, which determine the overall tuning of the EFM1 (see Adjusting Global EFM1 Parameters).

Some tuning ratio examples to try

- Set the modulator and carrier to the first harmonic (a 1:1 ratio), and the EFM1 will produce a sawtooth-like sound.
- Set the modulator to the second harmonic, and the carrier to the first harmonic (a 2:1 ratio), and you will produce a tone that sounds similar to a square wave.

Choosing a Different EFM1 Modulator Waveform

In classic FM synthesis, sine waves are used as modulator and carrier waveforms. The EFM1 Modulator oscillator provides a number of additional digital waveforms, which extend its sonic capabilities significantly. These waveforms contain a number of additional harmonics which add a new level of richness to the resulting FM sounds.

To choose a different waveform

- Turn the Wave parameter knob.
- At the full left position, the modulator produces a sine wave.
- As you turn the Wave parameter clockwise, you will step/fade through a series of complex digital waveforms.

Working with the EFM1 Modulation Parameters

FM synthesis is, at its core, caused by the intensity and type of modulations that take place in the signal path. Therefore, the “modulators” outlined in this section have a different impact (and role to play) than equivalent envelopes and LFOs found in analog synthesizer designs.

- *Modulation Env(elope) sliders:* Controls both the FM (Intensity) and Modulator pitch parameters over time. The envelope is triggered every time a MIDI note is received.
  - *Attack slider:* Sets the time it takes to reach the maximum envelope level.
  - *Decay slider:* Sets the time it takes to reach the Sustain level.
  - *Sustain slider:* Sets a level that is held until the MIDI note is released.
  - *Release slider:* Sets the time it takes to reach a level of zero, after the MIDI note has been released.
- *Modulator Pitch knob:* Determines the impact of the modulation envelope on the pitch of the modulator oscillator.
  - Turn the knob clockwise to increase the effect of the modulation envelope. Turn the knob counter-clockwise to invert the effect of the modulation envelope: the envelope slopes down during the attack phase, and slopes up during the decay and release time phases.
  - At the center (0) position, the envelope has no effect on the pitch of the Modulator oscillator. Click the “0” to center the Modulator Pitch knob.

- *FM Depth knob:* Determines the impact of the modulation envelope on FM intensity.
  - Turn the knob clockwise to increase the effect of the modulation envelope. Turn the knob counter-clockwise to invert the effect of the modulation envelope: the envelope slopes down during the attack phase, and slopes up during the decay and release time phases.
  - At the center (0) position, the envelope has no effect on FM intensity. Click the "0" to center the FM Depth knob.
- *LFO (Low Frequency Oscillator) knob:* Determines the amount of modulation applied to FM intensity or pitch.
  - Turn the LFO knob clockwise to increase the effect of the LFO on FM Intensity. Turn it counter-clockwise to introduce a vibrato.
  - At the center (0) position the LFO has no effect. Click the "0" to center the LFO knob.
- *Rate knob:* Sets the speed of the LFO.

Adjusting Global EFM1 Parameters

The global parameters are used to set the tuning, number of voices and other aspects of the EFM1's overall sound.

- *Transpose pop-up menu:* Sets the base pitch. You can transpose the EFM1 by octaves with this control.
- *Tune field:* Used to fine-tune the pitch of the EFM1 by cents. A cent is 1/100th of a semitone.
- *Voices pop-up menu:* Determines the number of simultaneously playable voices (polyphony). Click the pop-up menu to choose from mono (one voice), legato (one voice), or any number from 2 to 16 voices.

  Note: In the monophonic, legato mode, playing overlapping notes does not retrigger the EFM1 envelopes.
- *Unison button:* Turn on the Unison button to layer two complete EFM1 voices, thus making the sound larger and fatter. The EFM1 can be played with (up to) 8-voice polyphony when in unison mode.
- *Glide field:* Used to introduce a continuous pitch bend between two consecutively played notes. Adjust the Glide value (in ms) to determine the time it takes for the pitch to travel from the last played note to the next.

Note: Glide can be used in both of the monophonic (mono and legato) modes, or in any of the polyphonic settings (Voices parameter = 2 to 16).

Setting EFM1 Output Parameters

The EFM1 provides the following level controls.

- *Sub Osc Level knob:* Used for added bass response. The EFM1 features a sine wave sub oscillator. This operates one octave below the FM engine (as determined by the Transpose parameter). Turning up the Sub Osc Level control mixes the sub oscillator sine wave with the EFM1's FM engine output.
- *Stereo Detune knob:* Adds a rich and diverse chorus-like effect to the sound of the EFM1. This is achieved by doubling the EFM1 voice with a secondary, detuned FM engine. High values result in a wide stereo effect being added to the detuning, increasing the perceived space and width of your sound.

  Note: It is possible that mono compatibility can be lost with use of this parameter.
- *Vol(ume) envelope:* Shapes the level of the sound over time. The volume envelope is triggered every time a MIDI note is received.
  - *Attack slider:* Sets the time it takes to reach the maximum volume level.
  - *Decay slider:* Sets the time it takes to reach the Sustain level.
  - *Sustain slider:* Sets a level that is held until the MIDI note is released.
  - *Release slider:* Sets the time it takes to reach a level of zero, after the MIDI note has been released.
- *Main Level knob:* Sets the overall output level of the EFM1.
- *Velocity knob:* Determines the sensitivity of the EFM1 to incoming MIDI velocity messages. The EFM1 dynamically reacts to MIDI velocity messages-harder playing will result in a brighter and louder sound. Set the Velocity control all the way to the left (counter-clockwise) if you don't want the EFM1 to respond to velocity.

Creating EFM1 Random Sound Variations

The Randomize facility (shown at the lower right of the interface) generates new sounds. It does this by randomly altering a number of key EFM1 parameter values.

This feature is ideal for creating subtle variations of a particular sound, or for creating totally new sounds when you're feeling uninspired (or need a hand to get started with FM synthesis).

To use the Randomize function

- Click the Randomize button. You can click as often as you like, but remember to save your settings as you go, if you happen across a sound that you want to keep.

To limit the amount of randomization

- Drag in the numeric field to determine the amount of randomization (variance from the original sound).

  You should use values below 10% if you only want to randomly "tweak" the current sound.

  Use higher values if you want to radically change the sound with each click.

Assigning EFM1 MIDI Controllers

The EFM1 Extended Parameters area allows you remotely control the EFM1 with your MIDI controller keyboard (or other MIDI device). You may assign any unused (and suitable) MIDI controller to the following parameters:

- FM intensity
- Vibrato

To assign a controller

- Choose the desired controller in the Ctrl FM or Ctrl Vibrato menu, and set the modulation or vibrato amount with the slider below the menu.

  Note: The EFM1 also responds to MIDI pitch bend data: Pitch bend is hard-wired to the overall pitch of the EFM1.

EVOC 20 PolySynth

7

The EVOC 20 PolySynth combines a vocoder with a polyphonic synthesizer and can be played in real time.

It can create classic vocoder sounds, made famous by artists such as Kraftwerk during the 1970s and 1980s. Vocoding remains popular in current electronic, hip-hop, R & B, and other music styles.

This chapter covers the following:

- How the EVOC 20 Polysynth Works (p. 141)
- Getting to Know the EVOC 20 PolySynth Interface (p. 144)
- EVOC 20 PolySynth Sidechain Analysis Parameters (p. 145)
- EVOC 20 PolySynth (U/V) Detection Parameters (p. 147)
- EVOC 20 PolySynth Synthesis Section Parameters (p. 148)
- EVOC 20 PolySynth Formant Filter Parameters (p. 153)
- EVOC 20 PolySynth Modulation Parameters (p. 156)
- EVOC 20 PolySynth Output Parameters (p. 157)
- Getting the Best Results with the EVOC 20 PolySynth (p. 158)
- A Brief Vocoder History (p. 160)
- EVOC20 Block Diagram (p. 163)

How the EVOC 20 Polysynth Works

EVOC 20 PolySynth "listens" to an incoming audio signal—typically of a spoken or sung performance—and imposes the sonic characteristics and level changes of this signal onto the integrated synthesizer.

When you play notes and chords with your MIDI keyboard, the internal synthesizer will "sing" at the pitches of incoming MIDI notes, but with the articulations—level changes, vowel and consonant sounds—of the incoming audio signal.

This results in the classic “singing robot” or “synthetic voice” sounds that vocoders are mainly known for.

EVOC 20 PolySynth offers more than vocoding, however. You can use it as a synthesizer, or for more subtle effects processing—such as the creation of (somewhat) natural sounding vocal harmonies from a solo voice performance. If your musical tastes are more extreme, feel free to try processing other audio material, such as drum or instrument loops.

What Is a Vocoder?

The word *vocoder* is an abbreviation for *VOice enCODER*. A vocoder analyzes and transfers the sonic character of the audio signal arriving at its analysis input to the synthesizer’s sound generators. The result of this process is heard at the output of the vocoder.

The classic vocoder sound uses speech as the analysis signal and a synthesizer sound as the synthesis signal. This sound was popularized in the late 1970s and early 1980s. You’ll probably know it from tracks such as “O Superman” by Laurie Anderson, “Funky Town” by Lipps Inc., and numerous Kraftwerk pieces—such as “Autobahn,” “Europe Endless,” “The Robots,” and “Computer World.”

In addition to these “singing robot” sounds, vocoding has also been used in many films—such as with the Cylons in Battlestar Galactica, and most famously, with the voice of Darth Vader from the Star Wars saga. Also see A Brief Vocoder History.

Vocoding, as a process, is not strictly limited to vocal performances. You could use a drum loop as the analysis signal to shape a string ensemble sound arriving at the synthesis input.

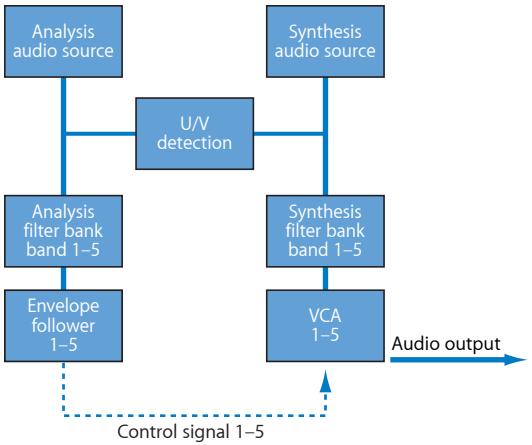
How Does a Vocoder Work?

The speech *analyzer* and *synthesizer* features of a vocoder are actually two bandpass *filter banks*. Bandpass filters allow a frequency band—a slice in the overall frequency spectrum—to pass through unchanged, and cut the frequencies that fall outside the band’s range.

In the EVOC 20 plug-ins, these filter banks are named the Analysis and Synthesis sections. Each filter bank has a matching number of corresponding bands—if the analysis filter bank has five bands (1, 2, 3, 4, and 5), there will be a corresponding set of five bands in the synthesis filter bank. Band 1 in the analysis bank is matched to band 1 in the synthesis bank, band 2 to band 2, and so on.

The audio signal arriving at the analysis input passes through the analysis filter bank, where it is divided into bands.

An envelope follower is coupled to each filter band. The envelope follower of each band tracks, or *follows*, any volume changes in the audio source—or, more specifically, the portion of the audio that has been allowed to pass by the associated bandpass filter. In this way, the envelope follower of each band generates dynamic control signals.



These control signals are then sent to the synthesis filter bank—where they control the levels of the corresponding synthesis filter bands. This is done with voltage-controlled amplifiers (VCAs) in analog vocoders. Volume changes to the bands in the analysis filter bank are imposed on the matching bands in the synthesis filter bank. These filter level changes are heard as a synthetic reproduction of the original input signal—or a mix of the two filter bank signals.

The more bands a vocoder offers, the more precisely the original sound's character will be reproduced by the synthesis filter bank. The EVOC 20 PolySynth provides up to 20 bands per bank. See EVOC20 Block Diagram for a detailed image of the EVOC20 Polysynth signal path.

Setting Up Your EVOC 20 PolySynth Host Application

To use of the EVOC 20 PolySynth, you need to insert it into the Instrument slot of an instrument channel strip. You also need to provide an audio signal as the analysis audio source, via a sidechain.

To set up the EVOC 20 PolySynth in your host application

1. Insert the EVOC 20 PolySynth into the Instrument slot of an instrument channel strip.
2. Choose an input source in the Side Chain menu in the plug-in header of the EVOC 20 PolySynth. This can be an audio track, live input, or bus, depending on the host application.

   The EVOC 20 PolySynth is now ready to accept incoming MIDI data and has been assigned to an input, audio track, or bus—via a side chain.

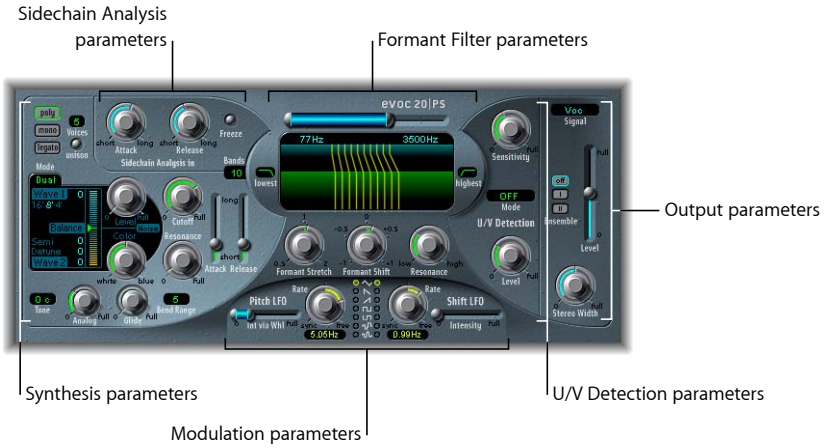
3 If applicable to your host application and needs, mute the audio track serving as the side chain input, start playback, and play your MIDI keyboard.

4 Adjust the volume levels of the EVOC 20 PolySynth and the Side Chain source (if not muted) to taste.

5 Experiment with the knobs, sliders, and other controls. Have fun, and feel free to insert other effect plug-ins to further enhance the sound.

Getting to Know the EVOC 20 PolySynth Interface

The EVOC 20 PolySynth interface is divided into six main parameter sections.



- *Sidechain Analysis parameters:* Determine how the input signal is analyzed and used by the EVOC 20 PolySynth. See EVOC 20 PolySynth Sidechain Analysis Parameters.
- *U/V Detection parameters:* Detect the unvoiced portions of the sound in the analysis signal, improving speech intelligibility. See EVOC 20 PolySynth (U/V) Detection Parameters.
- *Synthesis parameters:* Control the polyphonic synthesizer of the EVOC 20 PolySynth. See EVOC 20 PolySynth Synthesis Section Parameters.
- *Formant Filter parameters:* Configure the analysis and synthesis filter banks. See EVOC 20 PolySynth Formant Filter Parameters.
- *Modulation parameters:* The Modulation section offers two LFOs—used to modulate the synthesizer and filter banks. See EVOC 20 PolySynth Modulation Parameters.
- *Output parameters:* Configure the output signal of the EVOC 20 PolySynth. See EVOC 20 PolySynth Output Parameters.

EVOC 20 PolySynth Sidechain Analysis Parameters

The parameters in the Sidechain Analysis section control how the input signal is analyzed and used by the EVOC 20 PolySynth. You should be as precise as possible with these parameters to ensure the best possible speech intelligibility and accurate tracking.

- *Attack knob:* Determines how quickly each envelope follower—coupled to each analysis filter band—reacts to rising signal levels. Longer attack times result in a slower tracking response to transients—level spikes—of the analysis input signal. A long attack time on percussive input signals—a spoken word or hi-hat part, for example—will translate into a less articulated vocoder effect. Therefore, you should set the Attack parameter to the lowest possible value to enhance articulation.
- *Release knob:* Determines how quickly each envelope follower—coupled to each analysis filter band—reacts to falling signal levels. Longer release times cause the analysis input signal transients to sustain for a longer period at the vocoder's output. A long release time on percussive input signals—a spoken word or hi-hat part, for example—will translate into a less articulated vocoder effect. Use of extremely short release times results in rough, grainy vocoder sounds. Release values of around 8 to 10 ms are useful starting points
- *Freeze button:* This parameter, when enabled, holds—or *freezes*—the current analysis sound spectrum indefinitely. While Freeze is enabled, the analysis filter bank ignores the input source, and the Attack and Release knobs have no effect. See Freezing the EVOC 20 PolySynth's Input Signal.
- *Bands field:* Determines the number—up to 20—of frequency bands that are used by the filter banks. See Setting the Number of EVOC 20 PolySynth Filter Bank Bands.

Freezing the EVOC 20 PolySynth's Input Signal

The Freeze button in the Sidechain Analysis section of the EVOC20 PolySynth freezes the sound spectrum of the analysis input signal.

By freezing the input signal you can capture a particular characteristic of the signal, which is then imposed as a complex sustained filter shape on the Synthesis section. Here are some examples of when this could be useful:

- If you are using a spoken word pattern as a source, the Freeze button could capture the attack or tail phase of an individual word within the pattern—the vowel *a*, for example.
- If you want to compensate for people's inability to sustain sung notes for a long period, without taking a breath, you can use the Freeze button. If the synthesis signal needs to be sustained but the analysis source signal—a vocal part—is not sustained, use the Freeze button to lock the current formant levels of a sung note, even during gaps in the vocal part, between words in a vocal phrase. The Freeze parameter can be automated, which may be useful in this situation.

A Short Primer on Formants

A *formant* is a peak in the frequency spectrum of a sound. When the term is used in relation to human voices, formants are the key component that enables humans to distinguish between different vowel sounds—based purely on the frequency of these sounds. Formants in human speech and singing are produced by the vocal tract, with most vowel sounds containing four or more formants.

Setting the Number of EVOC 20 PolySynth Filter Bank Bands

The Bands field in the Sidechain Analysis section determines the number of frequency bands the EVOC 20 PolySynth's filter bank uses.

The greater the number of frequency bands, the more precisely the sound can be reshaped. As the number of bands is reduced, the source signal's frequency range is divided up into fewer bands—and the resulting sound will be formed with less precision by the synthesis engine. You may find that a good compromise between sonic precision—allowing incoming signals such as speech and vocals to remain intelligible—and resource usage is around 10 to 15 bands.

EVOC 20 PolySynth (U/V) Detection Parameters

Human speech consists of a series of voiced sounds—tonal sounds or formants—and unvoiced sounds. The main distinction between voiced and unvoiced sounds is that voiced sounds are produced by an oscillation of the vocal cords, whereas unvoiced sounds are produced by blocking and restricting the air flow with lips, tongue, palate, throat, and larynx.

If speech containing voiced and unvoiced sounds is used as a vocoder's analysis signal, but the synthesis engine doesn't differentiate between voiced and unvoiced sounds, the result will sound rather weak. To avoid this problem, the synthesis section of the vocoder must produce different sounds for the *voiced* and *unvoiced* parts of the signal.

The EVOC 20 PolySynth includes an Unvoiced/Voiced detector for this specific purpose. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with noise, with a mixture of noise and synthesizer signal, or with the original signal. If the U/V Detector detects voiced parts, it passes this information to the Synthesis section, which uses the normal synthesis signal for these portions.

- *Sensitivity knob:* Determines how responsive U/V detection is. When this knob is turned to the right, more of the individual unvoiced portions of the input signal are recognized. When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V sound source—determined by the Mode menu, as described in "Mode Menu" below—being used on the majority of the input signal, including voiced signals. Sonically, this results in a sound that resembles a radio signal that is breaking up and contains a lot of static, or noise.
- *Mode menu:* Sets the sound sources that can be used to replace the unvoiced content of the input signal. You can choose between the following:
  - *Noise:* Uses noise alone for the unvoiced portions of the sound.
  - *Noise + Synth:* Uses noise and the synthesizer for the unvoiced portions of the sound.

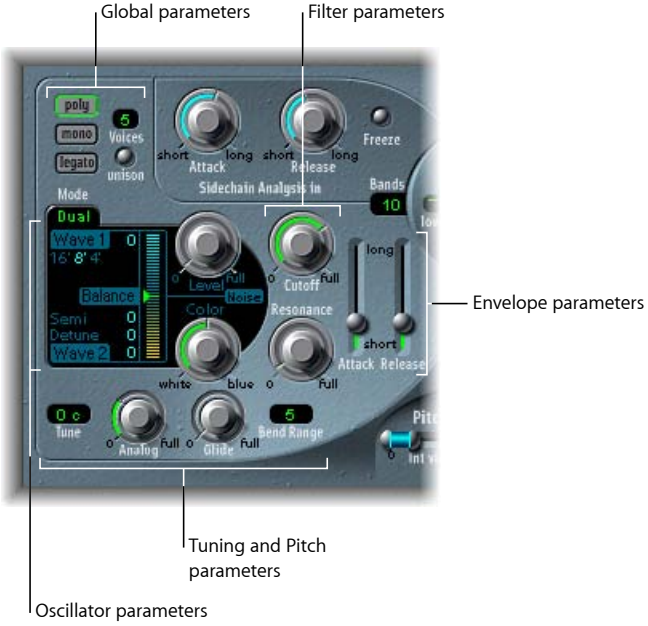
- *Blend:* Uses the analysis signal after it has passed through a highpass filter for the unvoiced portions of the sound. The Sensitivity parameter has no effect when this setting is used.

- *Level knob:* Controls the volume of the signal used to replace the *unvoiced* content of the input signal.

  Important: Take care with the Level knob, particularly when a high Sensitivity value is used, to avoid internally overloading the EVOC 20 PolySynth.

EVOC 20 PolySynth Synthesis Section Parameters

The EVOC 20 PolySynth is equipped with a polyphonic synthesizer. It is capable of accepting MIDI note input. The parameters of the Synthesis section are described below.



- *Oscillator parameters:* Used to choose the basic waveforms for the synthesis engine of the EVOC 20 PolySynth. See EVOC 20 PolySynth Oscillator Parameters.
- *Tuning and Pitch parameters:* Control the overall tuning of the synthesizer, and aspects such as pitch bend and portamento. See EVOC 20 PolySynth Tuning and Pitch Parameters.
- *Filter parameters:* Used to shape the basic waveforms of the oscillators. See EVOC 20 PolySynth Filter Parameters.
- *Envelope parameters:* Provide level control of the attack and release phases of the synthesizer sound. See EVOC 20 PolySynth Envelope Parameters.

- *Global parameters:* The parameters at the top left of the interface determine the keyboard mode and number of voices used by the EVOC 20 PolySynth. See EVOC 20 PolySynth Global Parameters.

EVOC 20 PolySynth Oscillator Parameters

The EVOC 20 PolySynth has two oscillators, which you can switch between Dual mode and FM mode:

- *Dual mode:* Each oscillator allows you to choose a digital waveform.
- *FM mode:* Oscillator 1 generates a sine wave. The frequency, or *pitch*, of Oscillator 1 is modulated by Oscillator 2 (for information about FM synthesis, see Frequency Modulation (FM) Synthesis). This leads to a number of different tones and harmonics becoming audible. Oscillator 2 can use any of the digital waveforms.

Each mode subtly changes the parameters shown in the Oscillator section.

The Synthesis section also incorporates a noise generator, which can add a further color to your sound. See the following for information on Dual mode, FM mode, and Noise generator mode:

Dual Mode EVOC 20 PolySynth Oscillator Parameters

FM Mode EVOC 20 PolySynth Oscillator Parameters

EVOC 20 PolySynth Noise Generator Parameters

Common EVOC 20 PolySynth Oscillator Parameters

This section describes parameters common to both Dual and FM modes.

- *16', 8', 4' value buttons:* Click to select the octave range for Oscillator 1. 16' (16 feet) is the lowest, and 4' the highest setting. The use of the term *feet* to determine octaves comes from the measurements of organ pipe lengths. The longer (and wider) the pipe, the deeper the tone.

- *Wave 1 and Wave 2 fields:* Vertically drag the numerical value beside the Wave 1 and Wave 2 labels to select the waveform type for Oscillators 1 and 2, respectively. The EVOC 20 PolySynth features 50 single-cycle digital waveforms with different sonic characteristics.

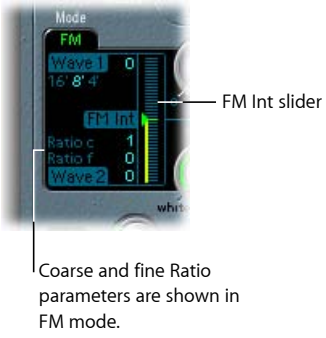
Dual Mode EVOC 20 PolySynth Oscillator Parameters

In Dual mode, each oscillator can use any of 50 digital waveforms.

- *Semi field:* Adjusts the tuning of Oscillator 2 in semitone steps.
- *Detune field:* Fine-tunes both oscillators in cents. 100 cents equals a semitone step.
- *Balance slider:* Sets the level balance between the two oscillator signals.

FM Mode EVOC 20 PolySynth Oscillator Parameters

When in FM mode, Oscillator 1 generates a sine wave. The Wave 1 parameter has no effect in this mode.



- *"Ratio c(oarse)" field:* Adjusts the frequency ratio between Oscillator 2 and Oscillator 1 in semitone steps.
- *"Ratio f(ine)" field:* Adjusts the frequency ratio between Oscillator 2 and Oscillator 1 in cents.
- *FM Int slider:* Determines the intensity of modulation. Higher values result in a more complex waveform with more overtones.

EVOC 20 PolySynth Noise Generator Parameters

The noise generator provides a further sound source that can be used alongside the two oscillators.

Important: The noise generator in the Oscillator section is independent of the noise generator in the U/V Detection area. For further information about voiced and unvoiced signals, see EVOC 20 PolySynth (U/V) Detection Parameters.

- *Level knob:* Controls the amount of noise added to the signals of the two oscillators.
- *Color knob:* Controls the timbre of the noise signal. At the full-left position, the noise generator creates pure white noise. At the full-right position, you hear blue noise (high-passed noise). White noise has always been used to create wind and rain sound effects. It has the same energy in each frequency interval. Blue noise sounds brighter, because its bass portion is suppressed by a highpass filter.

  Tip: Set Color to the full-right position and Level to a very low value to achieve a more lively and fresh synthesis signal.

EVOC 20 PolySynth Tuning and Pitch Parameters

The parameters at the lower left of the interface are used to control the overall tuning and other pitch-related aspects of the EVOC 20 PolySynth sound.

- *Analog knob:* Simulates the instability of analog circuitry found in vintage vocoders by randomly altering the pitch of each note. This behavior is much like that of polyphonic analog synthesizers. The Analog knob controls the intensity of this random detuning.
- *Tune field:* Defines the overall tuning of the EVOC 20 PolySynth, in cents.
- *Glide knob:* Determines the time it takes for the pitch to slide from one note to another—portamento. (For information about mono and legato, see EVOC 20 PolySynth Global Parameters.)
- *Bend Range field:* Determines the pitch bend modulation range, in semitone steps.

EVOC 20 PolySynth Filter Parameters

The EVOC 20 PolySynth synthesis section has a simple lowpass filter. The filter is used for rough signal shaping, before the signal is more precisely shaped by the individual bands of the Formant Filter banks.

- *Cutoff knob:* Sets the cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies are removed from the synthesizer signal.
- *Resonance knob:* Boosts or cuts the signal portion that surrounds the frequency defined by the Cutoff knob.

  Tip: Set cutoff as high as possible, and add a little resonance to achieve a nice, brilliant high-end.

EVOC 20 PolySynth Envelope Parameters

The EVOC 20 PolySynth features a simple Attack/Release envelope generator, which is used to control the level of the oscillators over time.

- *Attack slider:* Determines the amount of time it takes for the oscillators to reach their maximum level.
- *Release slider:* Determines the amount of time it takes for the oscillators to reach their minimum level, after the keys have been released.

EVOC 20 PolySynth Global Parameters

The parameters at the top left of the interface determine the keyboard mode and number of voices used by the EVOC 20 PolySynth.

- *Poly button/Voices field:* When Poly is selected, the maximum number of Voices can be set in the numeric field.
- *Mono/Legato buttons:* When Mono or Legato is selected, the EVOC 20 PolySynth is monophonic, and uses a single voice.
  - In Legato mode, Glide (see EVOC 20 PolySynth Tuning and Pitch Parameters) is only active on tied notes. Envelopes are not retriggered when tied notes are played (single trigger).
  - In Mono mode, Glide is always active and the envelopes are retriggered by every note played (multi trigger).
- *Unison button:* Enables or disables Unison mode.
  - In Unison/Poly mode—where both the Unison and Poly buttons are active—each EVOC 20 PolySynth voice is doubled, which cuts polyphony in half (to a maximum of 8 voices, shown in the Voices field). The doubled voices are detuned by the amount defined with the Analog knob.
  - In Unison/Mono mode—where both the Unison and Mono or Legato buttons are active—up to 16 voices can be stacked and played monophonically. The Voices field displays the number of stacked voices that will be heard.

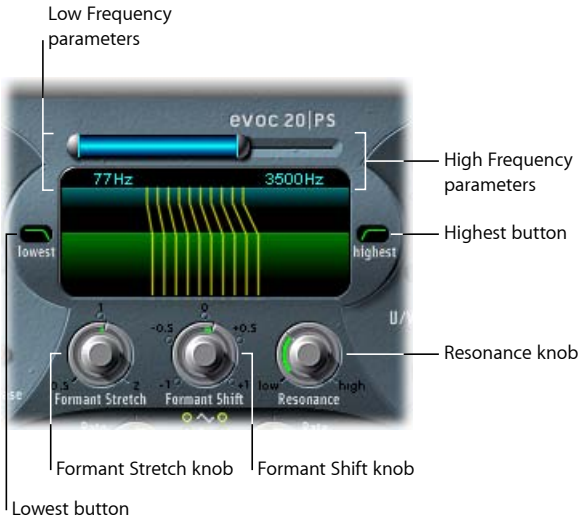
  Important: Stacking voices in Unison/Mono mode significantly increases the EVOC 20 PolySynth's output volume. To avoid overloading the instrument channel strip output, you should initially set a low Level slider value and gradually bring it up (see EVOC 20 PolySynth Output Parameters).

EVOC 20 PolySynth Formant Filter Parameters

The EVOC 20 PolySynth features two formant filter banks—one for the Analysis section and one for the Synthesis section. Each bank can house up to 20 individual filters. In essence, the entire frequency spectrum of an incoming signal is analyzed by the Analysis section and divided equally into a number of frequency bands. These analysis filter bands are mirrored by a corresponding number of bands in the synthesis filter bank. Each filter bank controls the peak levels—the formants—within these frequency bands. For more information, see How Does a Vocoder Work?, and A Short Primer on Formants.

EVOC 20 PolySynth Formant Filter Parameters Overview

The Formant Filter display is divided in two by a horizontal line. The upper half applies to the Analysis section and the lower half to the Synthesis section. Parameter changes are instantly reflected in the Formant Filter display, providing invaluable feedback about what is happening to the signal as it is routed through the two formant filter banks.



- *High and Low Frequency parameters:* Determine the lowest and highest frequencies allowed to pass by the filter section. Frequencies that fall outside these boundaries will be cut.
  - The length of the horizontal blue bar at the top represents the frequency range for both analysis and synthesis (unless Formant Stretch or Formant Shift are used, as described below). You can move the entire frequency range by dragging the blue bar. The silver handles on either end of the blue bar set the Low Frequency and High Frequency values, respectively.
  - You can also drag vertically in the numeric fields to adjust the Low and High frequency values.
- *Lowest and Highest buttons:* Click to determine whether the lowest and highest filter bands act as bandpass filters—like all of the bands between them—or whether they act as lowpass or highpass filters, respectively.
  - *Lowest button:* Click to determine whether the lowest filter band acts as a bandpass or highpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Highpass setting, all frequencies below the lowest bands are filtered.

- *Highest button:* Click to determine whether the lowest filter band acts as a bandpass or lowpass filter. In the Bandpass setting, the frequencies below the lowest bands and above the highest bands are ignored. In the Lowpass setting, all frequencies above the highest bands are filtered.

- *Formant Stretch knob:* Alters the width and distribution of all bands in the synthesis filter bank. This can be a broader or narrower frequency range than that defined by the Low and High Frequency parameters. See Using EVOC 20 PolySynth Formant Stretch and Shift.
- *Formant Shift knob:* Moves all bands in the synthesis filter bank up or down the frequency spectrum. See Using EVOC 20 PolySynth Formant Stretch and Shift.
- *Resonance knob:* Determines the basic sonic character of the vocoder. Low settings result in a softer character; high settings result in a more snarling, sharper character. Technically, increasing the Resonance value emphasizes the middle frequency of each frequency band.

Using EVOC 20 PolySynth Formant Stretch and Shift

When Formant Stretch is set to 0, the width and distribution of the bands in the synthesis filter bank at the bottom matches the width of the bands in the analysis filter bank at the top. Low values narrow the width of each band in the synthesis filter bank, whereas high values widen the bands. The control range is expressed as a ratio of the overall bandwidth.

When Formant Shift is set to 0, the positions of the bands in the synthesis filter bank match the positions of the bands in the analysis filter bank. Positive values move the synthesis filter bank bands up in frequency, whereas negative values move them down—in respect to the analysis filter bank band positions.

When combined, Formant Stretch and Formant Shift alter the formant structure of the resulting vocoder sound, which can lead to some interesting timbre changes. For example, using speech signals and tuning Formant Shift up results in “Mickey Mouse” effects.

Formant Stretch and Formant Shift are also useful if the frequency spectrum of the synthesis signal does not complement the frequency spectrum of the analysis signal. You could create a synthesis signal in the high frequency range from an analysis signal that mainly modulates the sound in a lower frequency range, for example.

Note: The use of the Formant Stretch and/or the Formant Shift parameters can result in the generation of unusual resonant frequencies, when high Resonance settings are used.

EVOC 20 PolySynth Modulation Parameters

The Modulation section offers two LFOs. The LFOs can run freely, or they can be synchronized with the host application tempo.

- The Pitch LFO controls pitch modulation of the oscillators, enabling you to produce vibrato effects.
- The Shift LFO controls the Formant Shift parameter of the synthesis filter bank, enabling you to produce dynamic phasing-like effects.

- *Int via Whl slider:* Defines the intensity of LFO pitch modulation. The right half of the slider determines the intensity when the modulation wheel is set to its maximum value, and the left half determines the intensity when the wheel is set to its minimum value. By dragging the area between the two slider segments, you can simultaneously move both. This parameter is permanently assigned to the modulation wheel of your MIDI keyboard (or corresponding MIDI data).
- *Rate knobs:* Determine the speed of modulation. Values to the left of the center positions are synchronized with the host application tempo and include bar values, triplet values, and more. Values to the right of the center positions are nonsynchronized, and are displayed in Hertz (cycles per second).

  Note: The ability to use synchronous bar values could be used to perform a formant shift every four bars on a cycled one-bar percussion part, for example. Alternatively, you could perform the same formant shift on every eighth-note triplet within the same part. Either method can generate interesting results and can lead to new ideas, or add life to existing audio material.
- *Waveform buttons:* Set the waveform type used by the Pitch LFO on the left or the Shift LFO on the right. You can choose between triangle, falling and rising sawtooth, square up and down around zero (bipolar, good for trills), square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S & H), and a smoothed random waveform for each LFO.
- *Intensity slider:* Controls the amount of formant shift modulation by the Shift LFO.

EVOC 20 PolySynth Output Parameters

The Output section provides control over the type, stereo width, and level of signal that is sent from the EVOC 20 PolySynth. The Output section also houses a simple, but effective, ensemble effect processor.

- *Signal menu:* Determines the signal that is sent to the EVOC 20 PolySynth main outputs. You can choose one of the following settings:
  - *Voc(oder):* Choose to hear the vocoder effect.
  - *Syn(thesis):* Choose to hear only the synthesizer signal.
  - *Ana(lysis):* Choose to hear only the analysis signal.

  Note: The last two settings are mainly useful for monitoring purposes.
- *Ensemble buttons:* Click to turn the ensemble effect on or off. Ensemble I is a special chorus effect. Ensemble II is a variation, creating a fuller and richer sound by using a more complex modulation routine.
- *Level slider:* Controls the volume of the EVOC 20 PolySynth output signal.
- *Stereo Width knob:* Distributes the output signals of the Synthesis section's filter bands in the stereo field.
  - At the left position, the outputs of all bands are centered.
  - At the centered position, the outputs of all bands ascend from left to right.
  - At the right position, the bands are output—alternately—to the left and right channels.

Getting the Best Results with the EVOC 20 PolySynth

Achieving a great "classic" vocoder effect requires both the analysis and synthesis signals to be of excellent quality, and it also requires care to be taken with the vocoder parameters. These sections outline a number of tips that will help you achieve the best possible results.

Editing the EVOC 20 PolySynth Analysis and Synthesis Signals

Avoiding Sonic Artifacts in the EVOC 20 PolySynth

FM Mode EVOC 20 PolySynth Oscillator Parameters

Editing the EVOC 20 PolySynth Analysis and Synthesis Signals

The following section outlines how you can edit the analysis and synthesis signals to achieve better speech intelligibility.

Compressing the EVOC 20 PolySynth Analysis Signal

The less the level changes, the better the intelligibility of the vocoder. You should therefore compress the analysis signal in most cases.

Enhancing High Frequency Energy

The vocoder, in a way, always generates the intersection point of the analysis and synthesis signals. If there's no treble portion in the analysis signal, the resulting vocoder output will also lack treble. This is also the case when the synthesis signal features a lot of high frequency content. Because this is true of each frequency band, the vocoder demands a stable level in *all* frequency bands from *both* input signals in order to obtain the best results.

Due to the way human beings hear, the intelligibility of speech is highly dependent on the presence of high frequency content. To aid in keeping speech clear, it may be worthwhile using equalization to boost or cut particular frequencies in analysis signals before you process them with the vocoders.

If the analysis signal consists of vocals or speech, a simple shelving filter should be sufficient to boost the high-mid and treble range, which is so important for speech intelligibility.

If the *synthesis* signal lacks treble energy, it can be generated with a distortion effect. The Logic Express Overdrive effect is perfect for this purpose.

Avoiding Sonic Artifacts in the EVOC 20 PolySynth

A common problem with vocoder sounds involves sudden signal interruptions (ripping, breaking sounds) and rapidly triggered noises during speech pauses.

Effective Use of the Envelope Parameters in the Analysis Section

The Release parameter defines the time it takes for a given synthesis frequency band to decrease in level if the signal level of the respective analysis band decreases abruptly. The sound is smoother when band levels decrease slowly. To achieve this smoother character, use higher Release values in the analysis section of the interface. Don't go too far with this, however, as overly long release times result in a less distinct, washy sound.

Short Attack values are no problem. They may, in fact, even be desirable when a fast reaction to incoming signals is required.

Gating Background Noises in the Analysis Signal

If the analysis signal is compressed, as recommended, the level of breath, rumble, and background noise will rise. These unwanted signals can cause the vocoder bands to open unintentionally. To eliminate these artifacts, you can employ a noise gate before using compression and boosting the treble frequencies. If the analysis signal is gated appropriately, you may be able to reduce the (Analysis) Release value.

When you are gating speech and vocals with the Logic Express Noise Gate plug-in, use Threshold to define the level above which the gate will open, and use Hysteresis to define a lower Threshold level, below which the gate will close. The Hysteresis value is relative to the Threshold level.

The figure above shows a Threshold setting that is well-suited for speech compression. Unwanted triggering by low or high frequency noise is avoided by the dedicated sidechain filters of the Noise Gate plug-in. The Hold, Release, and Hysteresis values are suitable for most vocal and speech signals.

Enhancing Speech Intelligibility in the EVOC 20 PolySynth

Keep these points in mind when trying to achieve the best possible speech intelligibility:

The spectra of the analysis and synthesis signals should almost completely overlap. Coupling low male voices with synthesis signals in the treble range doesn't work well.

The synthesis signal must be constantly sustained, without breaks. The incoming side chain signal should be played or sung legato, as breaks in the synthesis signal will stop the vocoder's output. Alternatively, the Release parameter of the synthesis signal—not the Release time of the analysis section—can be set to a longer time. You can also achieve nice effects by using a reverberation signal as a synthesis signal. Note that the two latter methods can lead to harmonic overlaps.

Do not overdrive the vocoder. This can happen easily, and distortion will occur.

Enunciate your speech clearly if the recording is to be used as an analysis signal. Spoken words, with a relatively low pitch, work better than sung vocals—even if the creation of vocoder choirs is your goal! Pronounce consonants well, as exemplified in the rolled *"R"* of "We are the Robots," by Kraftwerk, a classic vocoder track. This exaggerated pronunciation was specifically made to cater to the vocoder.

Feel free to do as you wish when setting the Formant parameters. Shifting, stretching, or compressing the formants has a surprisingly minimal effect on the intelligibility of speech. Even the number of frequency bands has a minimal impact on the quality of speech intelligibility.

The reason for this is our ability to intuitively differentiate the voices of children, women, and men, whose skulls and throats vary vastly by nature. Such physical differences cause variations in the formants that make up their voices. Our perception, or recognition, of speech is based on an analysis of the *relationships* between these formants. In the EVOC 20 plug-ins, these relationships are maintained even when extreme formant settings are used.

A Brief Vocoder History

You may be surprised to learn that the voder and vocoder date back to 1939 and 1940, respectively.

Homer Dudley, a research physicist at Bell Laboratories in New Jersey, developed the Voice Operated reCOrDER as a research machine. It was originally designed to test compression schemes for the secure transmission of voice signals over copper phone lines.

It was a composite device consisting of an analyzer and an artificial voice synthesizer, as follows:

- *Parallel bandpass vocoder:* A speech analyzer and resynthesizer, invented in 1940.

- *Vocoder speech synthesizer:* A voice modeler, invented in 1939. This valve-driven machine was played by a human operator. It had two keyboards, buttons to recreate consonants, a pedal for oscillator frequency control, and a wrist-bar to switch vowel sounds on and off.

The analyzer detected the energy levels of successive sound samples, measured over the entire audio frequency spectrum via a series of narrow band filters. The results of this analysis could be viewed graphically as functions of frequency against time.

The synthesizer reversed the process by scanning the data from the analyzer and supplying the results to a number of analytical filters, hooked up to a noise generator. This combination produced sounds.

The Voder was demonstrated at the 1939 World Fair, where it caused quite a stir. In World War II, the vocoder (known then as the VOice enCODER) proved to be of crucial importance, scrambling the transoceanic conversations between Winston Churchill and Franklin Delano Roosevelt.

Werner Meyer-Eppler, the director of Phonetics at Bonn University, recognized the relevance of the machines to electronic music—following a visit by Dudley in 1948. Meyer-Eppler used the vocoder as a basis for his future writings which, in turn, became the inspiration for the German "Elektronische Musik" movement.

In the 1950s, a handful of recordings ensued.

In 1960, the Siemens Synthesizer was developed in Munich. Among its many oscillators and filters, it included a valve-based vocoding circuit.

In 1967, a company called Sylvania created a number of digital machines that used time-based analysis of input signals, rather than bandpass filter analysis.

In 1971, after studying Dudley's unit, Bob Moog and Wendy Carlos modified a number of synthesizer modules to create their own vocoder for the *Clockwork Orange* sound track.

Peter Zinovieff's London-based company EMS developed a standalone—and altogether more portable—vocoder. EMS is probably best known for the Synthi AKS and VCS3 synthesizers. The EMS Studio Vocoder was the world's first commercially available machine, released in 1976. It was later renamed the EMS 5000. Among its users were Stevie Wonder and Kraftwerk. Stockhausen, the German "Elektronische Musik" pioneer, also used an EMS vocoder.

Sennheiser released the VMS 201 in 1977, and EMS released the EMS 2000, which was a cut-down version of its older sibling.

1978 saw the beginning of mainstream vocoder use, riding on the back of popularity created through the music of Herbie Hancock, Kraftwerk, and a handful of other artists. Among the manufacturers who jumped into vocoder production at this time are Synton/Bode, Electro-Harmonix, and Korg, with the VC-10.

In 1979, Roland released the VP 330 ensemble/vocoder keyboard.

The late 1970s and early 1980s were the heyday of the vocoder. Artists who used them included ELO, Pink Floyd, Eurythmics, Tangerine Dream, Telex, David Bowie, Kate Bush, and many more.

On the production side, vocoders could—and can still—be picked up cheaply in the form of kits from electronics stores.

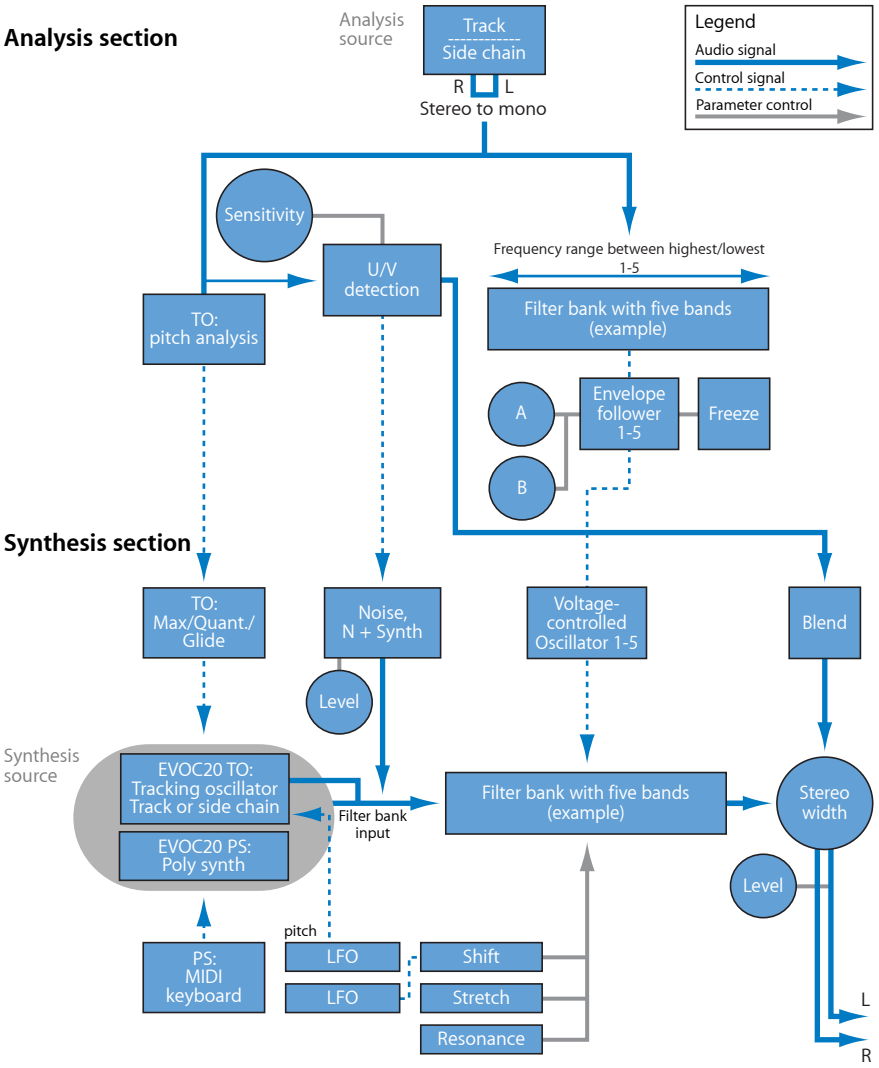
From 1980 to the present, EMS in the UK, Synton in Holland, and PAiA in the USA have been—and remain—the main flyers of the vocoding flag.

In 1996, Doepfer in Germany and Music and More joined the vocoder-producing fraternity.

From the late 1990s to the present, a number of standalone and integrated software-based vocoders—like the EVOC 20—have appeared.

EVOC20 Block Diagram

This block diagram illustrates the signal path in the EVOC 20 TrackOscillator and EVOC 20 PolySynth.



External Instrument

8

You can use the External Instrument to route external MIDI sound generators through the Logic Express Mixer, which you can then process with Logic Express effects.

You can also use the External Instrument to transmit and receive MIDI information through the instrument channel strip that it is inserted in. This enables you to control an external module—both MIDI and audio—from within one element.

To avoid constant repatching of devices, it is best to use an audio interface that supports multiple inputs and outputs. The External Instrument plug-in is inserted in instrument channel strips in place of a software instrument.

This chapter covers the following:

- Getting to Know the External Instrument Interface (p. 165)
- Using the External Instrument (p. 166)

Getting to Know the External Instrument Interface

This section will familiarize you with the parameters of the External Instrument. See Using the External Instrument for details on use.

- *MIDI Destination pop-up menu:* Choose the desired MIDI instrument and channel.
- *Input pop-up menu:* Choose the inputs of your audio hardware that the MIDI sound generator is connected to.
- *Input Volume slider and field:* Determines the incoming signal level.

Using the External Instrument

The track routed to an instrument channel strip, which is being used for an external MIDI sound module, behaves just like a standard software instrument track. This enables you to record and play back MIDI regions on it, with the following benefits:

- You can take advantage of the sounds and synthesis engine of your MIDI module, with no overhead on your Mac CPU—apart from effects used in the channel strip.
- You not only can use insert effects, but you also can use send effects, by routing the instrument channel strip to aux channel strips.
- You can bounce your external MIDI instrument parts—with or without effects—to an audio file, in real time. This makes the creation of a mix, inclusive of all internal and external devices and tracks, a one-step process.

Note: Bouncing an External Instrument track cannot happen faster than real time, as is the case with any bounce operations where MIDI hardware is involved.

When using multitimbral MIDI sound sources, maximum flexibility can be gained by using multiple External Instrument instances. In this situation, a separate audio output of the tone generator (if equipped with multiple outputs) would be connected to different inputs on your audio interface—each addressed by individual External Instruments.

To process external MIDI instruments with effects

1. Connect the output (or output pair) of your MIDI module with an input (or input pair) on your audio interface.

   Note: These can be either analog or digital connections if your audio interface and MIDI sound generator are equipped with either, or both.
2. Create an instrument channel strip.
3. Click the Instrument slot, and choose External Instrument from the pop-up menu.
4. Choose the MIDI Destination from the pop-up menu in the External Instrument window.
5. Open the Input pop-up menu and choose the input (of your audio interface) that the MIDI sound generator is connected to.
6. Adjust the Input Volume, if necessary.
7. Insert the desired effects in the Insert slots of the channel strip.

EXS24 mkII

9

EXS24 mkII is a software sampler. It plays back audio files, called *samples*, that you load into it. These samples are combined into tuned, organized collections called *sampler instruments*. As sampler instruments are based on audio recordings, they are ideally suited to emulating real instruments such as guitars, pianos, and drums.

This chapter covers the following:

- EXS24 mkII Features (p. 168)
- Getting to Know the EXS24 mkII Interface (p. 169)
- About EXS24 Sampler Instruments (p. 170)
- Getting to Know the EXS24 mkII Parameter Window (p. 172)
- Using the EXS24 mkII Sampler Instruments Pop-Up Menu (p. 173)
- Adjusting EXS24 mkII Global Parameters (p. 178)
- Using the EXS24 mkII Pitch Parameters (p. 182)
- Working with EXS24 mkII Filter Parameters (p. 184)
- EXS24 mkII Output Parameters (p. 187)
- Working with EXS24 mkII Modulation (p. 188)
- An Overview of the EXS24 mkII Instrument Editor (p. 202)
- Creating EXS24 mkII Instruments, Zones, and Groups (p. 205)
- Editing EXS24 mkII Zones and Groups (p. 210)
- Setting EXS24 mkII Zone Parameters (p. 213)
- Using the EXS24 mkII Zone Loop Parameters (p. 215)
- Setting EXS24 mkII Group Parameters (p. 216)
- Using the EXS24 mkII Advanced Group Selection Parameters (p. 218)
- Graphical Editing of EXS24 mkII Zones and Groups (p. 219)
- Saving, Renaming, and Exporting EXS24 mkII Instruments (p. 221)
- Editing Samples in the EXS24 mkII Sample Editor (p. 222)
- Using an External Instrument Editor with the EXS24 mkII (p. 223)

- Importing EXS24 mkII Sampler Instruments (p. 224)
- Managing EXS24 Sampler Instruments (p. 233)
- Setting EXS24 mkII Sampler Preferences (p. 234)
- Configuring EXS24 mkII Virtual Memory (p. 237)
- Advanced EXS24 mkII RAM Management (p. 239)
- Using the VSL Performance Tool in EXS24 mkII (p. 239)

EXS24 mkII Features

EXS24 mkII is used to play, edit, and create sampler instruments. You can assign the samples in sampler instruments to particular key and velocity ranges, and process them with the EXS24 mkII filters and modulators.

EXS24 mkII offers powerful modulation and editing features for an instrument of its type, and is a flexible synthesizer in its own right. This enables you to create expressive sounds by using any sample as a basic synthesizer "waveform."

EXS24 mkII can be used as a mono/stereo instrument, or you can route loaded samples to multiple outputs. This allows you to independently process individual drum sounds in a drum kit, for example.

Samples of almost unlimited length can be used in EXS24 mkII, by "streaming" them directly from a hard disk. This opens the door to many of the multigigabyte sample libraries available.

EXS24 mkII provides an extensive library of sampler instruments that offers piano, string, acoustic and electric guitar, drum, and many other sounds.

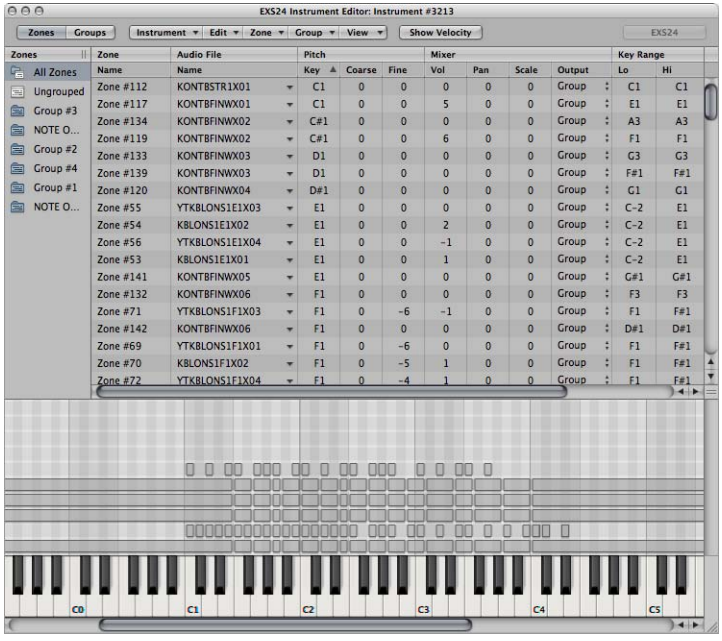
If you wish to extend your sonic repertoire, the EXS24 mkII native file format—the *EXS* format—is supported by most sample library providers. You can also import sampler instruments in the AKAI S1000 and S3000, SampleCell, Gigasampler, DLS, and SoundFont2 sample file formats.

Getting to Know the EXS24 mkII Interface

The EXS24 mkII interface is broken down into two windows:

- *Parameter window:* This is where you'll spend most of your time with the EXS24 mkII. It is used to load instruments, and offers a number of synthesis and modulation options that allow you to tailor your sampler instrument sounds.

- *Instrument Editor window:* Used to create and edit sampler instruments.



About EXS24 Sampler Instruments

A sampler instrument is the file type that is loaded into the EXS24 mkII. You load sampler instruments using the Sampler Instruments pop-up menu directly above the Cutoff knob in the EXS24 mkII. When you select a sampler instrument, the associated audio files are automatically located on the hard disk (or disks), and are loaded into your computer's RAM.

A sampler instrument tells the EXS24 mkII which samples—audio files—to use, and how to organize them into *zones* and *groups*. You play and record with the loaded sampler instrument in the same way as any software instrument. See An Overview of EXS24 mkII Zones and Groups and Differences Between EXS24 Sampler Instruments and Settings.

An Overview of EXS24 mkII Zones and Groups

A sampler instrument consists of zones and groups:

- A *zone* is a location into which a single sample—an audio file—can be loaded from a hard disk.
- Zones can be assigned to *groups*, which provide parameters that allow you to simultaneously edit all zones in the group. You can define as many groups as desired.

For information about zones and groups, see An Overview of the EXS24 mkII Instrument Editor and subsequent sections.

The EXS24 mkII is compatible with the following audio file formats: AIFF, WAV, SDII, and CAF. Each audio file is loaded into the EXS24 mkII as a separate sample. Each audio file is then automatically assigned to a *zone* in the EXS24 mkII Instrument Editor window. These zones can be edited and organized into sampler instruments. For more information about using audio files in zones, see Editing EXS24 mkII Zones and Groups.

Important: The actual audio files themselves are not contained within a sampler instrument. The sampler instrument only stores information about audio file names, parameter settings, and locations on the hard disk. If you delete or rename an audio file, any sampler instrument that makes use of this file will not be able to find it, so take care when performing these operations on audio files. You are free to move audio files to another location on your system, however. The EXS24 mkII can find moved files when sampler instruments are loaded.

Differences Between EXS24 Sampler Instruments and Settings

Sampler instruments are distinct from plug-in settings, which are loaded and saved in the plug-in window header. Each has advantages and disadvantages for handling parameter values in the Parameter window.

Typically, you will store the current Parameter window settings into the loaded sampler instrument. This overrides the settings currently saved in the loaded sampler instrument. Alternatively, you can save a new sampler instrument.

A plug-in setting, by comparison, stores all parameter adjustments made in the Parameter window, but these settings are discrete from the sampler instrument being loaded. A plug-in setting merely contains a pointer to an associated instrument, which means that loading a setting also loads the assigned sampler instrument.

So why have plug-in settings if you can store Parameter window values in sampler instruments?

The separation between plug-in settings and sampler instruments allows you to use sampler instruments as you would use waveforms in a synthesizer. For example, you could create a plug-in setting with guitar-like envelope, modulation, and filter parameter values. You would then use the Sampler Instruments pop-up menu to load an instrument (without any existing settings), such as a Flute, to create a plucked or strummed flute sound.

Important: Using sampler instruments as described requires that they contain no settings.

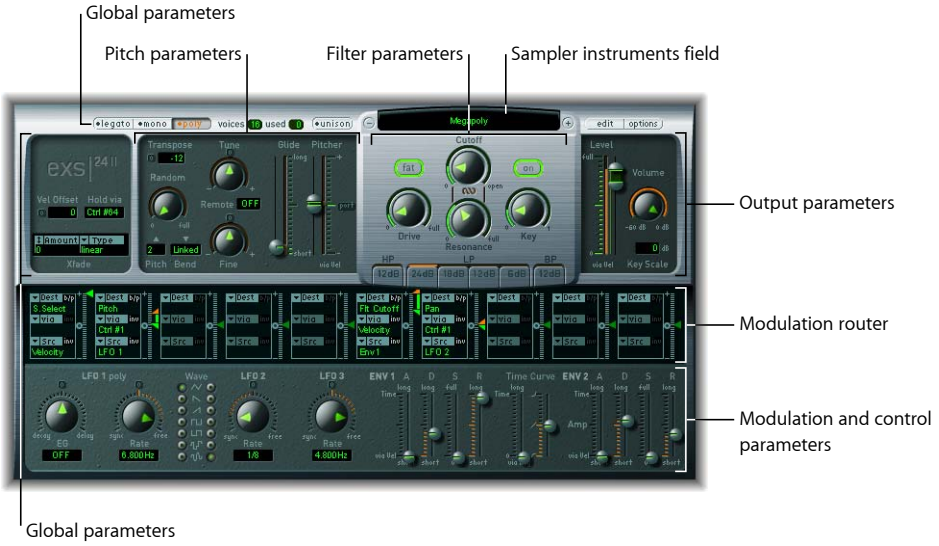
To remove settings from an existing sampler instrument

1. First, create a copy of the desired sampler instrument by using the Options > "Save instrument as" command (see Using the EXS24 mkII Options Pop-Up Menu Commands).
2. Remove the settings from the copied instrument by using the Options > "Delete settings from instrument" command.

Note: All of the sampler instruments supplied with Logic Express contain settings, so you need to follow the steps above if you wish to use these instruments as discussed.

Getting to Know the EXS24 mkII Parameter Window

The EXS24 mkII Parameter window is used to change and control the entire loaded sampler instrument. Control over individual samples (zones), or grouped samples, is performed in the Instrument Editor window (see An Overview of the EXS24 mkII Instrument Editor, and subsequent sections).



The following parameter groups are available:

- *Sampler Instruments pop-up menu and field:* Click to access and load your sampler instrument libraries. The name of the loaded sampler instrument is displayed in the field. The related Edit and Options buttons are found to the right. See Using the EXS24 mkII Sampler Instruments Pop-Up Menu.
- *Global parameters:* Used to select and configure sampler instruments, define the polyphony, set crossfades, and more. See Adjusting EXS24 mkII Global Parameters.
- *Pitch parameters:* Adjust tuning, transposition and pitch bend behavior. See Using the EXS24 mkII Pitch Parameters.
- *Filter parameters:* Used to shape the tonal color of the loaded sampler instrument. See Working with EXS24 mkII Filter Parameters.
- *Output parameters:* Used to control the level and keyboard scaling of the loaded sampler instrument. See EXS24 mkII Output Parameters.
- *Modulation router:* The strip across the center of the interface is the modulation router. The router links modulation sources, such as the envelopes and other parameters shown in the lower portion of the interface, to modulation destinations, such as the oscillators and filters. See Getting to Know the EXS24 mkII Modulation Router.

- *Modulation and control parameters:* The area immediately below the router is where you can assign and adjust the modulation and control parameters—the LFOs and envelopes. See Working with EXS24 mkII Modulation.

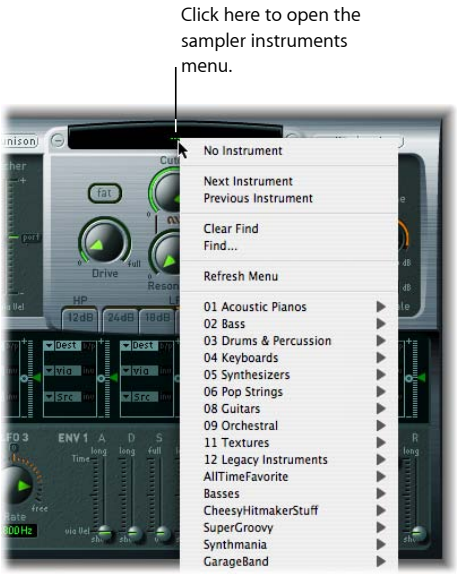
Using the EXS24 mkII Sampler Instruments Pop-Up Menu

The EXS24 mkII ships with a ready-to-play sampler instrument library. This section outlines the use of the Sampler Instruments pop-up menu. Also see Loading EXS24 Sampler Instruments from Other Locations and Searching for EXS24 Sampler Instruments.

The Edit and Options buttons, shown to the right of the Sampler Instruments field, are discussed in Opening the EXS24 mkII Instrument Editor and Using the EXS24 mkII Options Pop-Up Menu Commands.

To load an instrument

1. Click the Sampler Instruments field to open the Sampler Instruments pop-up menu.



2. Choose the desired sampler instrument.

About EXS24 Sample Storage Locations

In order to be visible in the EXS24 mkII Sampler Instruments pop-up menu, instruments must be stored in the Sampler Instruments sub-folder of any of the following folders:

- *~/Library/Application Support/Logic:* User-defined or edited instruments are stored here.
- */Library/Application Support/Logic:* Factory-supplied EXS instruments are installed here.
- */Applications/Logic 6 Series:* EXS instruments of the Logic Express 6 Series are stored here.
- *…/ProjectName:* Logic Express also searches for EXS Instruments in the project folder.

Note: You can store your sampler instruments in any folder on any of your computer's hard drives. Simply create an alias pointing to this folder within a Sampler Instruments sub-folder (in any of the paths listed above) and they will be shown in the Sampler Instruments pop-up menu.

To browse to the next or previous instrument in your sampler instrument library

Do one of the following:

- Click the plus or minus button on either side of the Sampler Instruments pop-up menu.

- Choose Next Instrument or Previous Instrument in the Sampler Instruments pop-up menu (or use the Next EXS Instrument and Previous EXS Instrument key commands).

  If the EXS24 mkII is the key focus window, you can also use the following key commands:

  - Next Plug-In Setting or EXS Instrument
  - Next Channel Strip or Plug-In Setting or EXS Instrument
  - Previous Plug-In Setting or EXS Instrument
  - Previous Channel Strip or Plug-In Setting or EXS Instrument

Tip: You can also browse through your sampler instruments by using your MIDI keyboard. The Sampler Preferences window offers Previous EXS Instrument and Next EXS Instrument options. These allow you to assign a MIDI event, such as a MIDI note, control change, program change, and so on, to select the previous or next sampler instrument in the Sampler Instruments pop-up menu. See Setting EXS24 mkII Sampler Preferences.

Loading EXS24 Sampler Instruments from Other Locations

You can manually load sampler instruments that are not shown in the Sampler Instruments pop-up menu. This is done from the Instrument pop-up menu in the Instrument Editor window.

To load sampler instruments from other locations

1. Open the Instrument Editor by clicking the Edit button in the upper-right corner of the Parameter window.

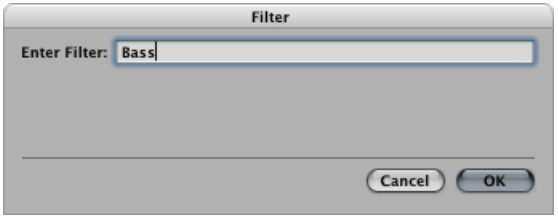
2. Choose Instrument > Open, then locate the desired instrument in the dialog.

Searching for EXS24 Sampler Instruments

To minimize the number of sampler instruments displayed in the Sampler Instruments pop-up menu, you can use the Find function. This limits the Sampler Instruments pop-up menu to displaying only sampler instrument names that contain the search term.

To search for sampler instruments

1. Click to open the Sampler Instruments pop-up menu, then choose Find.
2. Enter the search term in the Filter window.



To disable the Search Filter

- Choose Clear Find in the Sampler Instruments pop-up menu.

The full Sampler Instruments pop-up menu is displayed, but the search term you typed into the Filter window is not cleared. You can return to the limited menu by choosing Enable Find in the Sampler Instruments pop-up menu. This allows you to switch between the two views without re-typing the search term.

To perform a different search

- Choose the Find command again and enter the desired search term.

Opening the EXS24 mkII Instrument Editor

Click the Edit button to the right of the Sampler Instruments pop-up menu to open the currently loaded sampler instrument in the EXS24 mkII Instrument Editor window. The Instrument Editor window provides precise control over each sample, or zone, in the sampler instrument.

Note: Clicking the Edit button when no sampler instrument is loaded also opens the Instrument Editor window and automatically creates a new, empty, sampler instrument.

For detailed information, see An Overview of the EXS24 mkII Instrument Editor, and subsequent sections.

Using the EXS24 mkII Options Pop-Up Menu Commands

Click the Options button in the upper-right corner of the Parameter window. This opens the Options pop-up menu, which contains the following commands:

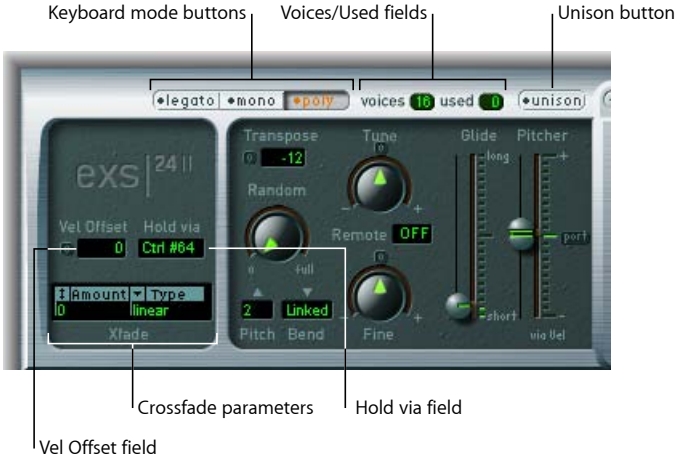
- *Recall default EXS24 settings:* Recalls a neutral setting for all parameters in the Parameter window. This provides a "clean slate" when adjusting the parameters of your sampler instrument.
- *Recall settings from instrument:* Recalls the original parameter settings of the loaded sampler instrument. This parameter is useful if you've been overzealous with your tweaking and want to return to the original sampler instrument parameter settings.
- *Save settings to instrument:* Stores the current parameter values of the Parameter window in the sampler instrument file. When the instrument is reloaded, these values are recalled.
- *Delete settings from instrument:* Removes the stored settings (Parameter window values) from the sampler instrument file.
- *Rename instrument:* Opens a file save dialog, allowing you to rename the loaded instrument. This overwrites the existing instrument name.
- *Save instrument as:* Opens a file save dialog, allowing you to enter an alternative name for the loaded instrument. This preserves the original name and sampler instrument file, and creates a new sampler instrument—a copy.

  Note: This is a good, safe option to use, as any projects or templates that use the originally named sampler instrument will work as expected.
- *Delete instrument:* Deletes the loaded sampler instrument.

- *(Recall default EXS24 mkI settings):* Recalls the parameter settings of sampler instruments created in the older version of the EXS24—notably, the modulation paths (see EXS24 mkI Modulation Paths). This parameter is not relevant for sampler instruments created in the EXS24 mkII.
- *Extract MIDI Region(s) from ReCycle Instrument:* Extracts regions contained in a ReCycle instrument. If no ReCycle instrument is selected, this option is dimmed. See Converting ReCycle Files to EXS Instruments.
- *AKAI Convert:* Launches the AKAI Convert window (see Converting AKAI Files with the EXS24 mkII).
- *SoundFont Convert, SampleCell Convert, DLS Convert, Giga Convert:* Each of these commands launches a dialog with instructions about performing these conversions. See Importing SoundFont2, SampleCell, DLS, and Gigasampler Files for details.
- *Preferences:* Opens the EXS24 mkII preferences (see Setting EXS24 mkII Sampler Preferences).
- *Virtual Memory:* Opens a configuration window for the EXS24 mkII's virtual memory functions. Virtual memory allows samples of almost unlimited length to be played back—by streaming audio directly from the hard disk in real time. For more information, see Configuring EXS24 mkII Virtual Memory. The virtual memory window also allows you to enable direct access to system memory for the EXS24 mkII—in systems with 5 GB of RAM or more. See Advanced EXS24 mkII RAM Management.

Adjusting EXS24 mkII Global Parameters

These parameters affect the overall behavior of the EXS24 mkII. You can find the global parameters at the top left of the interface.



- *Keyboard mode buttons:* Switch the EXS24 mkII between polyphonic, monophonic, and legato behaviors. See Choosing the EXS24 mkII Keyboard Mode.
- *Unison button:* Enables or disables unison mode. See Using Unison Mode in the EXS24 mkII.
- *Voices/Used fields:* The "voices" field determines the maximum number of notes that can be played simultaneously. The "used" field is a real-time monitor that indicates the number of voices that are actually used when you play the keyboard. See Setting the EXS24 mkII Voices Parameter.
- *Vel Offset field:* Increases or decreases the incoming MIDI note velocity value by ±127, expanding or limiting the dynamic response of the EXS24 mkII to incoming MIDI note events.
- *Hold via field:* Determines the modulation source used to trigger the sustain pedal function (hold all currently played notes, and ignore note-off messages until the modulation source value falls below 64). The default is MIDI controller number CC 64 (the standard MIDI "Hold/Sustain" controller number).
- *Crossfade parameters:* Allow you to crossfade between layered samples (zones)—with adjacent velocity ranges. See Using the EXS24 mkII Crossfade (Xfade) Parameters.

Choosing the EXS24 mkII Keyboard Mode

A *polyphonic* instrument allows several notes to be played simultaneously—for example, an organ or piano. Brass or reed instruments are *monophonic*, which means that only one note can be played at a time. The EXS24 mkII allows you to choose an appropriate keyboard mode for the type of instrument that is loaded. You are free to use a monophonic mode for polyphonic instruments, which allows playing styles that are not possible with polyphonic instruments.

- If you choose Mono mode, staccato playing will retrigger the envelope generators every time a new note is played. If you play in a legato style (play a new key while holding another), the envelope generators are triggered only for the first note you play legato and then continue their curve until you release the last legato played key.
- The Legato mode is also monophonic, but with one difference—the envelope generators are only retriggered if you play staccato (release each key before playing a new key).

Glide Behavior in Different Keyboard Modes

In Legato mode, Glide is active only on tied notes. Envelopes are not retriggered when tied notes are played—in other words, playing a series of tied notes results in only a single envelope trigger. For more information on the Glide function, see Using the EXS24 mkII Pitch Parameters.

In Mono mode, Glide is always active, and the envelopes are retriggered by every note played.

Using Unison Mode in the EXS24 mkII

In Unison mode, multiple EXS24 mkII voices are played when a key is struck. This enables a richer sound, achieved by slightly detuning each voice. This is ideal when emulating classic analog synthesizers.

To enable monophonic unison mode

- Activate either Mono or Legato mode and also turn on the Unison button:
  - The intensity of the unison effect depends on the number chosen in the Voices parameter field. Increase the Voices value for a fatter sound.

- The intensity of detuning (voice deviation) is set with the Random parameter (see Using the EXS24 mkII Pitch Parameters).

To use the EXS24 mkII in polyphonic unison mode

- Activate the Poly and Unison buttons.
  - In Poly/Unison mode, each played note is effectively doubled—or, more correctly, the polyphony value of the Voices parameter is halved. These two voices are then heard when you trigger the note. Activating Poly/Unison has the same effect as setting the EXS24 mkII to Mono/Unison (Voices = 2), but you can play polyphonically.

Voices are equally distributed in the panorama field and are evenly detuned. The Random knob value determines the amount of tuning deviation between voices.

Note: The number of voices actually used per note increases with the number of layered sample zones.

Setting the EXS24 mkII Voices Parameter

This parameter determines the maximum number of voices (polyphony) that the EXS24 mkII can play. The "used" field is a real-time monitor that indicates the number of voices that are actually used when you play the keyboard. If both fields tend to show the same value most of the time (probably causing a noticeable number of voices to drop out), you should set a higher *voices* value.

Using the EXS24 mkII Crossfade (Xfade) Parameters

The Xfade parameters allow you to crossfade between layered samples—known as *zones* in the EXS24 mkII—with adjacent velocity ranges. If you are unfamiliar with the concept of layering samples, see Layering EXS24 mkII Zones.

- *Amount field:* Expands the velocity range of all zones by applying an identical value to each layered zone. The crossfade takes place in the extended velocity range area. When the Amount parameter is set to 0, the EXS24 mkII will simply switch from one zone to another.

Note: You can also set other modulation sources, such as the modulation wheel of your MIDI keyboard, to modulate the Amount parameter. If you do this, the Amount parameter still functions in the same way, but the crossfade will be triggered by the modulation wheel rather than by velocity.

- *Type pop-up menu:* Provides three different fade curves. Choose the desired curve type for your velocity crossfade:
  - *dB lin (dB linear):* A logarithmic curve that evenly crossfades between zones
  - *linear (gain linear):* A convex crossfade curve with a rapid volume fade towards the end
  - *Eq. Pow (equal power):* A nonlinear curve with a rapid level increase at the beginning of the fade. This is useful if your crossfade seems to drop in volume part way through.

Layering EXS24 mkII Zones

When assigning a sample to a zone, you can set the lowest and highest MIDI note velocity that will trigger that zone. The area between these values is known as the zone's velocity range. You can layer zones—different samples—on the same keyboard note, and trigger them individually by playing at different velocities.

For example: Imagine you have layered two samples (zone 1 and zone 2) on MIDI note A#2.

- Zone 1 is a sample of a snare drum hit lightly, and a little off-center. It has a MIDI note velocity range of 24 to 90.
- Zone 2 is a sample of a snare drum, hit hard in the center of the drum head. It has a velocity range of 91 to 127.

As you can see, the maximum velocity range value of zone 1 and the minimum velocity range value of zone 2 are adjacent. If you were to play note A#2 at velocities above or below a value of 90, you would clearly hear each sample being triggered. To make this transition less abrupt, you can use the crossfade parameters to smoothly fade between each zone. Where you have distinctly different audio samples in adjacent zones, you will find crossfading very helpful in creating realistic-sounding sampler instruments.

Using the EXS24 mkII Pitch Parameters

These parameters adjust the tuning and transposition of the loaded sampler instrument.

- *Tune knob:* Raises or lowers the pitch of the loaded sampler instrument in semitone increments. At the centered position—set by clicking the small 0 button—no pitch change occurs.
- *Transpose field:* Transposes the EXS2 mkII in semitone increments. Transpose not only affects the pitch, but also moves the zones by the specified value.
- *Random knob:* Controls the amount of random detuning that is applied to every played voice. You can use Random to simulate the tuning drift of analog synthesizers, or to thicken the sound—very effective when emulating some stringed instruments.
- *Fine knob:* Tunes the loaded sampler instrument in cent increments (1/100th of a semitone). Use this to correct samples that are slightly out of tune, or to create a thick chorus-like effect.
- *Pitch Bend Up and Down pop-up menus:* Determine the upper and lower limit of pitch bending (in semitones) that can be introduced by moving your keyboard's pitch bend wheel. A value of 0 disables pitch bends.

  Note: When you choose Link mode in the right-hand Pitch Bend Down menu, the bend range is identical in both directions—if you assign an upward bend of 4 semitones, the downward bend will also be set to 4 semitones, resulting in a combined bend range of 8 semitones (9, if you include the standard pitch, or "no bend" position).
- *Remote field:* Used to change the pitch of complete EXS24 mkII instruments in real time. You can define a key on your MIDI keyboard that is used as the original, or "reference" pitch. Once set, playing any of the keys in a range of ±1 octave above or below this key will change the pitch of the entire instrument, rather than triggering the sample. This is somewhat similar to the Pitch Bend function, but is quantized to semitones.

- *Glide and Pitcher sliders:* The Glide slider determines the time it takes to slide from one note pitch to another. Its behavior depends on the Pitcher parameter setting:
  - When Pitcher is centered, Glide determines the time it takes for the pitch to slide from one note to another—the *portamento* time.
  - When the Pitcher parameter is set to a position above its centered value, Glide determines the time it takes for the pitch to glide down from this higher value to the normal pitch value.
  - When Pitcher is set to a position below the centered value, the pitch glides from this lower setting back up to the normal pitch value.

  The Pitcher parameter can be modulated by velocity—the upper half of the slider determines the setting for maximum velocity, the lower half for minimum velocity. By dragging the area between the two slider segments, you can move both simultaneously.

  When the upper half of the Pitcher slider is set above the center position and the lower half below the center position, lower velocity values cause the pitch to rise from the lower setting up to the original note pitch, whereas higher values cause it to fall from the higher setting down to the original note pitch. In other words, the polarity of the pitch envelope can be changed by velocity.

  When both halves of the pitcher slider are set either below or above the centered position, a low or high velocity will slide up or down to the original pitch. Depending on the position of the upper/lower halves of the slider—relative to the center position—the time required for the slide up or down to the original note pitch can be adjusted independently for both soft and hard velocities.

Working with EXS24 mkII Filter Parameters

These parameters control the EXS24 mkII filter section. You can configure the type of filter, filter resonance, cutoff frequency, drive, and amount of key follow. For details about the filter envelope, see Getting to Know the EXS24 mkII Envelopes (ENV 1 and ENV 2).

- *Filter On/Off button:* Activates or deactivates the entire filter section and the filter envelope. Deactivating the filter section makes it easier to hear adjustments to other sound parameters, as the filter always heavily affects the sound. Disabling the filter also reduces processor load. If the button is green and labeled On, the filter is engaged. If it's gray and labeled Off, the filter is disabled.
- *Filter Mode/Slope buttons:* Appearing as HP, LP, and BP at the bottom of the section, these buttons determine the type, and slope, of the filter. See Choosing the EXS24 mkII Filter Mode (HP, LP, BP).
- *Cutoff knob:* Sets the cutoff frequency of the filter. The *Cutoff* value also serves as the starting point for any modulation involving the filter. See Using EXS24 mkII Filter Cutoff and Resonance Parameters.
- *Resonance knob:* Boosts or cuts the frequency area surrounding the cutoff frequency. Very high Resonance values introduce self-oscillation, causing the filter to produce a sound—a sine wave—on its own. See Using EXS24 mkII Filter Cutoff and Resonance Parameters.
- *Drive knob:* Overdrives the filter input. Turning Drive up leads to a denser, more saturated signal, which introduces additional harmonics. See Overdriving the EXS24 mkII Filter.
- *Key knob:* Defines the amount of filter cutoff frequency modulation by note number. When Key is set to the full-left position, the cutoff frequency is unaffected by the note number and is identical for all played notes. When Key is set to the full-right position, the cutoff frequency follows the note number 1:1—if you play one octave higher, the cutoff frequency is also shifted upward by one octave. This parameter is useful for avoiding overly filtered high notes.

- *Fat (Fatness) button:* Enables or disables the fatness feature. Fatness preserves the bass frequency response of the loaded sampler instrument, even when high Resonance settings are used.

  Note: The Fatness parameter applies only to lowpass filters. Fatness is nonfunctional when the highpass or bandpass filter types are active.

Choosing the EXS24 mkII Filter Mode (HP, LP, BP)

The EXS24 mkII filter can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.

Select one of the following buttons at the bottom of the filter section to choose a filter mode:

- *HP (highpass):* This filter type allows frequencies above the cutoff frequency to pass. The slope of the highpass filter is fixed at 12 dB/octave in HP mode.
- *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. Click any of the four buttons under the LP label to engage the lowpass filter, and choose a filter slope: 24 dB (4 pole), 18 dB (3 pole), 12 dB (2 pole), and 6 dB (1 pole). The 24 dB setting can be used for drastic effects, such as cutting off all but a few notes. The 6 dB setting is useful for creating a slightly warmer sound, without drastic filter effects—to smooth overly bright samples, for example.
- *BP (bandpass):* The frequency band directly surrounding the cutoff frequency is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the center frequency of the band.

Using EXS24 mkII Filter Cutoff and Resonance Parameters

The following section outlines the impact of the filter's Cutoff and Resonance parameters. If you're new to synthesizers, and the concepts behind filters, see Filters in Synthesizer Basics.

The Impact of Cutoff Frequency on the Signal

The Cutoff Frequency (Cut) parameter controls the brilliance of the signal.

- In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
- In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed, with only upper frequencies allowed to pass.
- In a bandpass/band-rejection filter, the cutoff frequency determines the center frequency for the bandpass or band-rejection filter.

The Impact of Resonance on the Signal

The Resonance (Res) parameter emphasizes or suppresses portions of the signal above or below the defined cutoff frequency.

- In a lowpass filter, resonance emphasizes or suppresses signals below the cutoff frequency.
- In a highpass filter, resonance emphasizes or suppresses signals above the cutoff frequency.
- In bandpass filters, resonance emphasizes or suppresses the portions of the signal—the frequency band—that surround the defined frequency, set with the Cutoff Frequency parameter. Alternatively, resonance can be used to define the width of the frequency band. This is how it is used in the EXS24 mkII.

Controlling Cutoff and Resonance Simultaneously

Changing the Cutoff and Resonance controls at the same time is key to the creation of expressive synthesizer sounds.

To control two filter parameters at once

- Drag the chain symbol between the Cutoff and the Resonance knobs to control both parameters simultaneously—vertical movements alter Cutoff values, and horizontal movements affect Resonance values.

Overdriving the EXS24 mkII Filter

The filter is equipped with an overdrive module. Overdrive intensity is defined by the Drive parameter.

The filter Drive parameter affects each voice independently. When every voice is overdriven individually—like having six fuzz boxes for a guitar, one for each string—you can play extremely complex harmonies over the entire keyboard range. They'll sound clean, without unwanted intermodulation effects spoiling the sound.

Furthermore, appropriate Drive parameter settings lead to a different tonal character. The way analog filters behave when overdriven forms an essential part of the sonic character of a synthesizer. Each synthesizer model is unique in the way its filters behave when overdriven. The EXS24 mkII is quite flexible in this area, allowing tonal colors that range from the most subtle fuzz to the hardest of distortions.

EXS24 mkII Output Parameters

The output parameters define the level—the perceived volume—of a played note. The change in level over time is controlled by an *envelope generator*.

ENV2 is hard-wired to the dynamic stage of the EXS24 mkII—it is always used to control the level of each note. For a description of all the envelope parameters, see Getting to Know the EXS24 mkII Envelopes (ENV 1 and ENV 2).

- *Level via Vel slider:* Determines how velocity affects the volume of the sound. The upper half of the slider determines the volume when the keyboard is struck at maximum velocity, and the lower half determines the volume when the keyboard is struck at minimum velocity. Drag the area between the two slider segments to move both simultaneously.
- *Volume knob:* Acts as the main volume parameter for the EXS24 mkII. Adjust to find the balance between no distortion and getting the best—the highest—resolution in the channel strip and the Level via Vel slider.
- *Key Scale field:* Modulates the sampler instrument level by note number—position on the keyboard. Negative values increase the level of lower notes. Positive values increase the level of higher notes. This is useful when emulating a number of acoustic instruments, where higher-pitched notes are often louder than low notes.

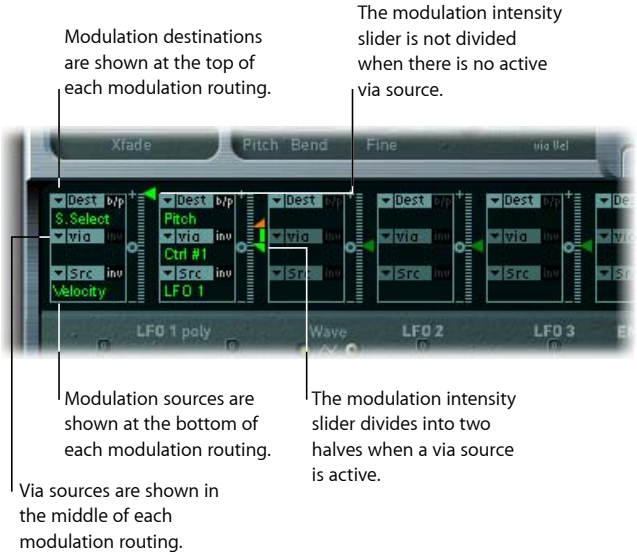
Working with EXS24 mkII Modulation

The EXS24 mkII is equipped with an extensive range of modulation sources and destinations, making it a very flexible instrument that can generate extraordinary sounds that constantly evolve, or are just plain expressive to play. Reference tables that cover all modulation destinations and sources are found at the end of this section.

- *Modulation router:* The modulation router—abbreviated to router—links modulation sources, such as the envelope, to modulation destinations, such as the filter. The router features ten modulation routings—arranged into columns. See Getting to Know the EXS24 mkII Modulation Router.
- *Modulation and control parameters:* These include the LFOs and envelopes. See Getting to Know the EXS24 mkII LFOs and Getting to Know the EXS24 mkII Envelopes (ENV 1 and ENV 2).

Getting to Know the EXS24 mkII Modulation Router

The modulation router spans the center of the EXS24 mkII interface. If you are new to synthesizer modulation routings, see Modulation Routing in Synthesizer Basics. Also see An EXS24 mkII Modulation Example.


Modulation destinations are shown at the top of each modulation routing.
The modulation intensity slider is not divided when there is no active via source.
Xfade
Pitch Bend
Fine
Dest
S.Select
Via
Src
Velocity
Pitch
Ctrl #1
LFO 1
LFO 1 poly
Wave
LFO 2
LFO 3
Modulation sources are shown at the bottom of each modulation routing.
The modulation intensity slider divides into two halves when a via source is active.
Via sources are shown in the middle of each modulation routing.

Any modulation *source* can be connected to any modulation *destination*—much like an old-fashioned telephone exchange or a studio patch bay. See Creating and Bypassing EXS24 mkII Modulation Routings, EXS24 mkII Modulation Source Reference, and EXS24 mkII Modulation Destination Reference.

The modulation intensity—how strongly the destination is influenced by the source—is set with the vertical slider to the right of the modulation routing.

The intensity of the modulation can itself be modulated: The *via* parameter defines a further modulation source, which is used to control the modulation intensity. When via is active, you can specify upper and lower limits for the modulation intensity. See Using EXS24 mkII Via Sources to Control Modulation Intensity and EXS24 mkII Modulation Via Source Reference.

Ten such modulation routings of source, via, and destination can take place simultaneously, in addition to routings that are hard-wired outside of the router. It does not matter which of the ten modulation routings you use.

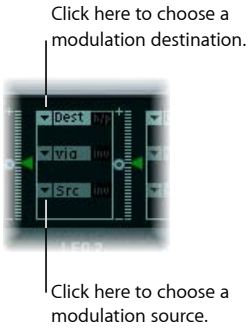
You can even select the same destination in several parallel modulation routings. You can also use the same sources and the same via controllers in multiple modulation routings.

Creating and Bypassing EXS24 mkII Modulation Routings

The following information applies to all ten modulation routings.

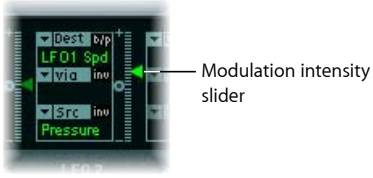
To create a basic modulation routing

1 Open the Dest pop-up menu to see all available destinations, and choose the parameter you want to modulate.



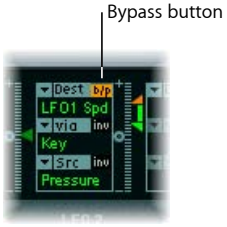
2 Open the Src pop-up menu to see all available sources, and choose the parameter you want to use to modulate the destination.

3 Vertically drag the arrowhead of the Intensity slider to the right of the modulation routing to set a fixed modulation intensity.



To bypass a modulation routing

- Click the “b/p” button at the top right of the modulation routing.



The Bypass (b/p) parameter allows you to enable or disable individual modulation routings, without losing settings.

Using EXS24 mkII Via Sources to Control Modulation Intensity

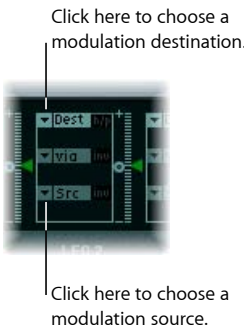
In a basic modulation routing consisting of a destination and source, you can set a fixed modulation intensity by vertically dragging the arrowhead of the Intensity slider to the right of the routing. The slider value always defines a constant modulation intensity.

The intensity of the modulation can itself be modulated: The *via* parameter defines a further modulation source, which is used to control the modulation intensity. As soon as you choose a value other than off for "via," the Intensity slider is divided into two halves, each half with its own arrowhead.

- The lower half defines the minimum modulation intensity, when the via controller—the modulation wheel, for example—is set to its minimum value.
- The upper half defines the maximum modulation intensity when the via controller is set to its maximum value.
- The area between the two slider halves defines the modulation range of the via controller.

To create a modulation routing that includes a via source

1. Open the Dest pop-up menu to see all available destinations, and choose the parameter you want to modulate.



2. Open the Src pop-up menu to see all available sources, and choose the parameter you want to use to modulate the destination.
3. Open the via pop-up menu to see all available sources, and choose the source you want to use for control of the modulation intensity.

4 Vertically drag the upper arrowhead of the Intensity slider, to the right of the modulation routing, to set the maximum modulation intensity.

5 Vertically drag the lower arrowhead of the Intensity slider to set the minimum modulation intensity.

To move the entire via range

- Drag the range area between the two slider halves vertically.

Both arrowheads will move simultaneously.

If this area is too small to drag, just drag an unused section of the Intensity slider control to move the area.

To set the modulation intensity to zero

- Click the small 0 symbol that is halfway up the Intensity slider control.

To invert the effect of the via modulation source

- Click the “inv” button to the right of the via pop-up menu.

An EXS24 mkII Modulation Example

The following example is useful for string sound modulations, where playing higher notes results in a faster modulation.

LFO1 Speed is the modulation destination.

The modulation source—Pressure—is used to modulate the speed (Rate) of LFO 1.

You’ll hear a faster modulation as you apply more pressure to the keyboard—after the initial keystrike.

The modulation intensity is controlled by keyboard position, the range of which is determined by the via parameter. In other words, the note (number) that you play controls the depth of LFO speed modulation.

EXS24 mkI Modulation Paths

Many of the hard-wired modulation paths that were available as sliders on the original EXS24 (mkI) are now integrated into the modulation router. To reconstitute the modulation slider configuration of the mkI version, open the Options pop-up menu in the upper-right corner of the interface and choose "(Recall Default EXS24 mkI Settings)." This will load the mkI modulation paths into the modulation router, as follows:

- Velocity to Sample Select
- LFO 1 to Pitch via ModWheel (= Ctrl#1)
- Velocity to Sample Start (inv)
- LFO 2 to Filter Cutoff via ModWheel
- Velocity to Filter Cutoff
- Envelope 1 to Filter Cutoff via Velocity
- LFO 2 to Pan via ModWheel

You can, of course, freely alter the settings of these modulation paths—allowing you to use modulation sources that were unavailable in the EXS24 mkI, for example (see the end of this section for tables of all sources and destinations).

Note: For technical reasons, the settings of the modulation router cannot translate backward to the EXS24 mkI.

Getting to Know the EXS24 mkII LFOs

The EXS24 mkII includes three LFOs (low frequency oscillators) that can be used as modulation sources. They are all available as sources or destinations in the router. If you are new to synthesizers, and the concept behind LFOs, see Using the LFO to Modulate Sounds.

LFO 1 is polyphonic, which means that if it is used for any modulation of multiple voices, they will *not* be phase-locked. Furthermore, LFO 1 is key-synced: Each time you play a key, the LFO 1 modulation of this voice is started from zero.

To understand the non-phase-locked characteristic more fully, imagine a scenario where a chord is played on the keyboard. If LFO 1 is used to modulate pitch, for example, the pitch of one voice might rise, the pitch of another voice might fall, and the pitch of a third voice might reach its minimum value. As you can see, the modulation is independent for each voice, or note.

The key sync facility ensures that the LFO waveform cycle always starts from zero, which results in consistent modulation of each voice. If the LFO waveform cycles were not synchronized in this way, individual note modulations would be uneven.

LFO 1 can also be faded in or out automatically, courtesy of a built-in envelope generator.

LFO 2 is monophonic, which means that the modulation is identical for all voices. To understand this more fully, imagine a scenario where a chord is played on the keyboard. If LFO 2 is used to modulate pitch, for example, the pitch of all voices in the played chord will rise and fall synchronously.

LFO 3 is also monophonic. It always uses a triangular waveform.

All three LFOs can oscillate freely, or can be synchronized to the host application tempo, in values ranging between 32 bars and 1/128 triplets.

- *LFO 1 EG knob:* Controls the time it takes for the LFO modulation to fade in or fade out (see Using the EXS24 mkII Envelope Generator of LFO 1).
- *LFO 1 Rate knob:* Defines the frequency—the speed—of the LFO 1 modulation. The value is displayed in Hertz (Hz) or in note values beneath the slider.
- *LFO 1 and 2 Wave buttons:* This is where you choose the desired waveform for LFO 1 and LFO 2. For details about how to use them, see Using EXS24 mkII LFO Waveforms.
- *LFO 2 Rate knob:* This parameter defines the frequency—the speed—of the LFO 2 modulation. See Setting the EXS24 mkII LFO Rate.
- *LFO 3 Rate knob:* This parameter defines the frequency—the speed—of the LFO 3 modulation. See Setting the EXS24 mkII LFO Rate.

Using EXS24 mkII LFO Waveforms

The Wave buttons allow you to choose different waveforms for LFO 1 and LFO 2. The table below outlines how these can affect your sounds.

Tip: Try different waveforms while a modulation routing of Pitch is engaged and running.

Waveform	Comments
Triangle	Well suited for vibrato effects
Sawtooth	Well suited for helicopter and space gun sounds. Intense modulations of pitch with a negative (inverse) sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects. The waveform can also be inverted, resulting in a different start point for the modulation cycle.

Waveform	Comments
Rectangle	Use of the rectangular waves will periodically switch the LFO between two values. The upper rectangular wave switches between a positive value and zero. The lower wave switches between a positive and a negative value set to the same amount above/below zero.An interesting effect you may want to try out is achieved by modulating the Pitch destination with a suitable modulation intensity that leads to an interval of a fifth. Choose the upper rectangular wave to do so.
Sample & Hold	The two lower waveform settings of the LFOs output *random* values. A random value is selected at regular intervals, as defined by the LFO rate. The upper waveform steps between randomized values (rapid switches between values). At its lower setting, the random wave is smoothed out, resulting in fluid changes to values. The term *Sample & Hold* (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then *held* until the next *sample* is taken.*Tip:* A random modulation of Pitch leads to an effect commonly referred to as a *random pitch pattern generator* or *sample and hold*. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies!

Using the EXS24 mkII Envelope Generator of LFO 1

LFO 1 features a simple envelope generator that is used to control the time it takes for the LFO modulation to fade in or fade out. At its center position, which can be accessed by clicking the middle mark, the modulation intensity is static—no fade-in or fade-out will occur.

To set the LFO 1 modulation fade time

- Choose a positive LFO 1 EG knob value to *fade in* the modulation.

  The higher the value, the longer the delay time.

- Choose a negative LFO 1 EG value to *fade out* the modulation.

  The lower the value, the shorter the fade-out time.

LFO envelopes are most often used for delayed vibrato—many instrumentalists and singers intonate longer notes this way.

To set up a delayed vibrato

1. Place the LFO 1 EG knob at a position towards the right (Delay) and modulate the Pitch destination with the LFO1 source in the router.
2. Set a slight modulation intensity.
3. Select an LFO 1 Rate of about 5 Hz.
4. Choose the triangular wave as the LFO 1 waveform.

Tip: Chaotic and fast modulations of frequencies (destination: Pitch) by the LFO 1 source—with a delayed Sample & Hold waveform, a high Rate, and short fade-out—are ideal for emulating the attack phase of brass instruments.

Setting the EXS24 mkII LFO Rate

LFO 2 is ideally suited for creating rhythmic modulation effects—effects that retain perfect synchronicity, even during project tempo changes. LFO 3 is much the same, but uses a fixed triangle waveform, making it most suitable for adding vibrato to a sound, or for use as a modulation source for the other LFOs.

The Rate parameter of all three LFOs allows the respective LFO to run freely (in the right-hand side of the Rate knob range), or to be synchronized with the project tempo (in the left-hand side of the Rate knob range).

The rate is displayed in Hertz or rhythmic values—the latter when project tempo synchronization is active. Rates range from speeds of 1/64-notes to a periodic duration of 32 bars. Triplet and punctuated values are also available.

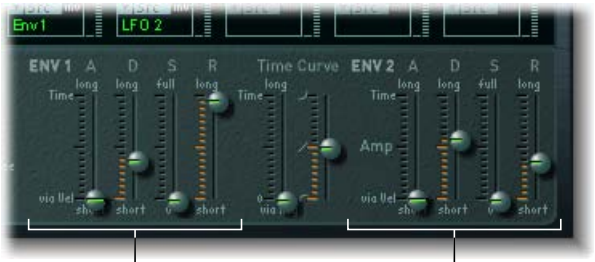
Getting to Know the EXS24 mkII Envelopes (ENV 1 and ENV 2)

The EXS24 mkII features two envelope generators per voice. They are abbreviated as ENV 1 and ENV 2 in the interface and router. For details on the roots of the term *envelope generator* and its basic functionality, see Envelopes in the Amplifier Section.

The parameters of ENV 1 and ENV 2 are identical.

- ENV 1 controls the filter over time.
- ENV 2 defines the changes in level over time—for each note played.

Both envelopes, however, are also available for simultaneous use as sources in the router. The envelope time parameters (Attack, Decay, and Release) are also available as modulation destinations in the router.



- *A(ttack) slider:* Defines the time it takes for the level of a note to rise from an amplitude of zero to the set amplitude. The Attack time sliders of both envelopes are divided into two halves.
  - The lower half defines the attack time when the keys are struck hard, at maximum velocity. The upper half defines the attack time at minimum velocity. Drag the area between the two slider halves to move them both simultaneously. If this area is too small to drag, drag an unused portion of the slider control.
- *D(ecay) slider:* Determines the time it takes for the level of a held note to fall to the sustain level after the attack phase has completed.
  - If the Sustain level parameter is set to its maximum value, the Decay parameter has no effect.
  - When the Sustain level is set to its minimum value, the Decay parameter defines the duration or fade-out time of the note.
- *S(ustain) slider:* Controls the sustain level.
- *R(elease) slider:* Defines the time required for the (sustain) level to decay to zero—after the key is released.
- *Time Curve sliders:* These apply to both envelopes. The left slider, known as *time via key*, can be used to scale (lengthen or shorten) the envelope time intervals. Note that position C3 is the center point.
  - Time intervals for zones assigned to keys above C3 can be reduced in length with the left slider. All time intervals for zones assigned to keys below C3 can be lengthened.
  - The (Attack) Curve slider determines the shape of the envelope attack curve.

EXS24 mkII Modulation Destination Reference

The following destinations are available for real-time modulation.

Destination	Comments
Sample Select	Modulates the sample (zone) that is played. By default, Sample Select is controlled by velocity—through the default Velocity to Sample Select modulation routing. This means that the received note velocity value determines which of the layered zones (in different velocity ranges) is heard as you play the keyboard softer or harder. You aren't limited to using velocity, however, to determine which sample is played. You could assign the modulation wheel source to the Sample Select destination, or use both velocity and the modulation wheel. If you choose a continuous controller such as the modulation wheel, you can step through the velocity layers during playback. In this case, make use of the crossfade (Xfade) parameters to create smooth transitions between velocity split points. When using multiple modulation sources, be aware that these can cause all velocity layers to run simultaneously—using up as many voices as there are layered zones. The CPU usage will increase accordingly.
Sample Start	Modulates the sample start time. This allows you to trigger a drum loop, partway through, for example.
Glide Time	Modulates the duration of the Glide (portamento) effect. If you modulate Glide, with Velocity selected as the source, the speed of the keystrike determines the time it takes for the played notes to reach the target pitch.
Pitch	Modulates the frequency (pitch) of the loaded sampler instrument. If you select an LFO as the source, this destination leads to siren or vibrato sounds. Select one of the envelope generators with zero attack, short decay, zero sustain, and short release as the source for tom and kick drum sounds. Slight envelope modulations can make the amount of detuning change over time, which can be particularly useful for brass sounds.
Filter Drive	Modulates the filter Drive parameter.
Filter Cutoff	Modulates the Cutoff Frequency parameter. See Working with EXS24 mkII Filter Parameters.
Filter Resonance	Modulates the Resonance parameter of the filter.
Volume	Controls the main output level of the EXS24 mkII.
Pan	Modulates the panorama position of the sound in the stereo spectrum. Modulating Pan with an LFO will result in a stereo tremolo (auto panning). In unison mode, the panorama positions of all voices are spread across the entire stereo spectrum. Nevertheless, pan can still be modulated, with positions being moved in parallel.
Relative Volume	Adds/subtracts the specified amount to/from the Volume parameter.

Destination	Comments
LFO 1 Dcy./Dly (LFO 1 Decay/Delay)	Controls the LFO 1 EG parameter (see Using the EXS24 mkII Envelope Generator of LFO 1).
LFO 1 Speed	Modulates the frequency (rate) of LFO 1. You can automatically accelerate/slow down LFO 1's rate by modulating the LFO1 Speed destination with one of the envelope generators (ENV) or with LFO2 or LFO 3.
LFO 2 Speed	As above, for LFO 2
LFO 3 Speed	As above, for LFO 3
Env 1 Attack	Modulates the Attack time of the filter envelope.
Env 1 Decay	Modulates the Decay time of the filter envelope.
Env 1 Release	Modulates the Release time of the filter envelope.
Time	Modulates the time via key slider position—see Time Curve sliders description in Getting to Know the EXS24 mkII Envelopes (ENV 1 and ENV 2).
Env 2 Attack (Amp)	Modulates the Attack time of the second envelope generator.
Env 2 Decay (Amp)	Modulates the Decay time of the second envelope generator. In cases where you've selected Env 2 Decay as the destination and Velocity as the source, the duration of the decaying note is dependent on how hard you strike the key. Selecting Key(board) as the source will result in higher notes decaying more quickly (or slowly).
Env 2 Release (Amp)	Modulates the Release time of the second envelope generator.
Hold	Modulates the (alternate) controller assigned to the sustain pedal function. See the Hold parameter information in Adjusting EXS24 mkII Global Parameters.

EXS24 mkII Modulation Source Reference

The following modulation sources are available:

Source	Comments
Side Chain	Side Chain modulation uses a side chain signal as a modulation signal. The side chain source can be selected in the Side Chain menu in the header of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.
Maximum	Max sets the value of this *source* to +1 (an internal value that indicates the maximum possible amount for this source). This offers interesting options for controlling the modulation intensity with all possible *via* values.
ENV 1	Envelope Generator 1 is used as a source.
ENV 2 (Amp)	Envelope Generator 2 is used as a source. Env 2 always controls the level of the overall sound.

Source	Comments
LFO 1	LFO 1 is used as a source.
LFO 2	As above, but for LFO 2
LFO 3	As above, but for LFO 3
Release Velocity	The modulation occurs when you release a key (this requires a keyboard that sends release velocity information).
Pressure	Pressure (also known as Aftertouch) serves as a modulation source. The EXS24 mkII reacts to poly pressure (polyphonic aftertouch).*Note:* If you set the *destination* to Cutoff, the cutoff frequencies will rise and fall, dependent on how firmly you press a key on your touch-sensitive MIDI keyboard—after the initial keystrike.
Pitch Bend	The pitch bend wheel is used as a modulation source.
Key	Kybd (Keyboard) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0, used internally by the EXS24 mkII). An output value of –1 indicates five octaves below (the center point). An output value of +1 indicates five octaves above. Modulate the Cutoff destination with the Key source to control the cutoff frequencies of the filter with the keyboard position—as you play up and down the keyboard, the cutoff frequencies change. A modulation intensity of 0.5 proportionately scales cutoff frequencies with keyboard note pitches.
Velocity	Velocity serves as a modulation source.
---	Disables the source.
MIDI Controllers 1–120	The chosen MIDI controller serves as a modulation source. Controllers 7 and 10 are marked as (not available). Host applications such as Logic Express use these controllers for volume and pan automation of channel strips. Controller 11 is marked as (Expression). It has a fixed connection to this functionality, but it can also be used to control other modulation sources.

EXS24 mkII Modulation Via Source Reference

The following *sources* may be used to control the modulation intensity.

via Source	Comments
Side Chain	Side Chain modulation uses a side chain signal as a modulation intensity (trigger) signal. The side chain source can be selected in the Side Chain menu in the header of the plug-in window. It is fed to the internal envelope follower, which creates a modulation value based on the current side chain input signal level.
Maximum	Sets the value of this *source* to +1.
ENV 1	Envelope Generator 1 controls the modulation intensity.
ENV 2 (Amp)	Envelope Generator 2 controls the modulation intensity.

via Source	Comments
LFO 1	The modulation undulates at the speed and waveform of LFO 1, which controls the modulation intensity.
LFO 2	As above, but for LFO 2
LFO 3	As above, but for LFO 3
Release Velocity	The modulation will be more or less intense dependent on how quickly you release the key (this requires a keyboard that sends release velocity information).
Pressure	If you select Pressure (also known as Aftertouch) as the via value, the modulation intensity will be touch sensitive—modulation will be more or less intense dependent on how firmly you press the key of your touch-sensitive MIDI keyboard after the initial keystrike.
Pitch Bend	The pitch bend wheel controls the modulation intensity.
Key	Key(board) outputs the keyboard position (the MIDI note number). The center point is C3 (an output value of 0). Five octaves below and above, an output value of –1 or +1, respectively is sent. If you select Pitch as the destination, modulate it with the LFO1 source, and select Key as the via value, the vibrato depth will change, dependent on key position. Put another way, the vibrato depth will be different for notes higher or lower than the defined Key(board) position.
Velocity	The modulation intensity will be velocity sensitive—modulation will be more or less intense depending on how quickly (hard) you strike the key.
---	Disables the via source.
MIDI Controllers 1–120	Modulation intensity is determined by the chosen MIDI controller value. Controllers 7 and 10 are marked as (not available). Host applications such as Logic Express use these controllers for volume and pan automation of channel strips. Controller 11 is marked as (Expression). It has a fixed connection to this functionality, but it can also be used to control other modulation sources.

An Overview of the EXS24 mkII Instrument Editor

The Instrument Editor is used to create and edit sampler instruments. See Creating EXS24 mkII Instruments, Zones, and Groups and Editing EXS24 mkII Zones and Groups.

A sampler instrument consists of zones and groups:

- A *zone* is a location into which a single sample (an audio file) is loaded from a hard disk. You can edit zone parameters in Zone view mode. See Getting to Know the EXS24 mkII Zones View Mode.

- Zones can be assigned to *groups*, which provide parameters that allow you to simultaneously edit all zones in the group. You can define as many groups as desired. You can edit group parameters in Group view mode. See Getting to Know the EXS24 mkII Groups View Mode.

To open the EXS24 mkII Instrument Editor

- Click the Edit button in the upper-right corner of the EXS24 mkII Parameter window.

Note: If no sampler instrument is loaded when you click the Edit button, a new instrument is automatically created.

The Instrument Editor has two view modes: Zones view and Groups view. Zones view displays zones and associated parameters, in the parameter area. Groups view shows groups and associated parameters.

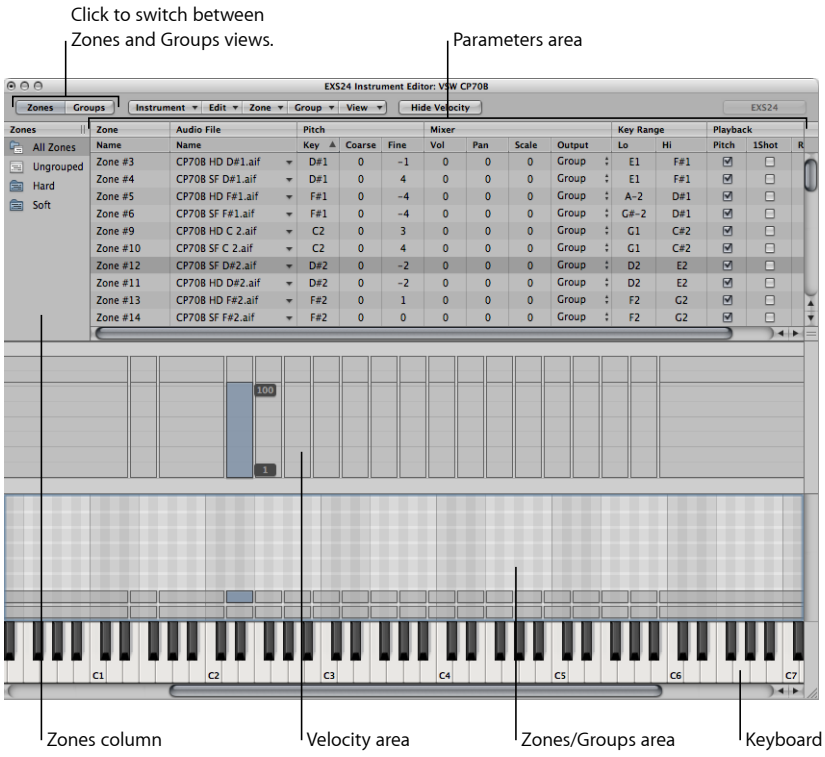
To switch between EXS24 mkII Instrument Editor view modes

- Click the Groups button in the upper-left corner to switch to Groups view, or click the Zones button in the upper-left corner to switch to Zones view.

  You can also use the Toggle Zones/Groups View key command to switch between views.

Getting to Know the EXS24 mkII Zones View Mode

In Zones view, the area above the keyboard displays the Zones area. The general menus, buttons, and so on, are displayed in both Zones and Groups views.

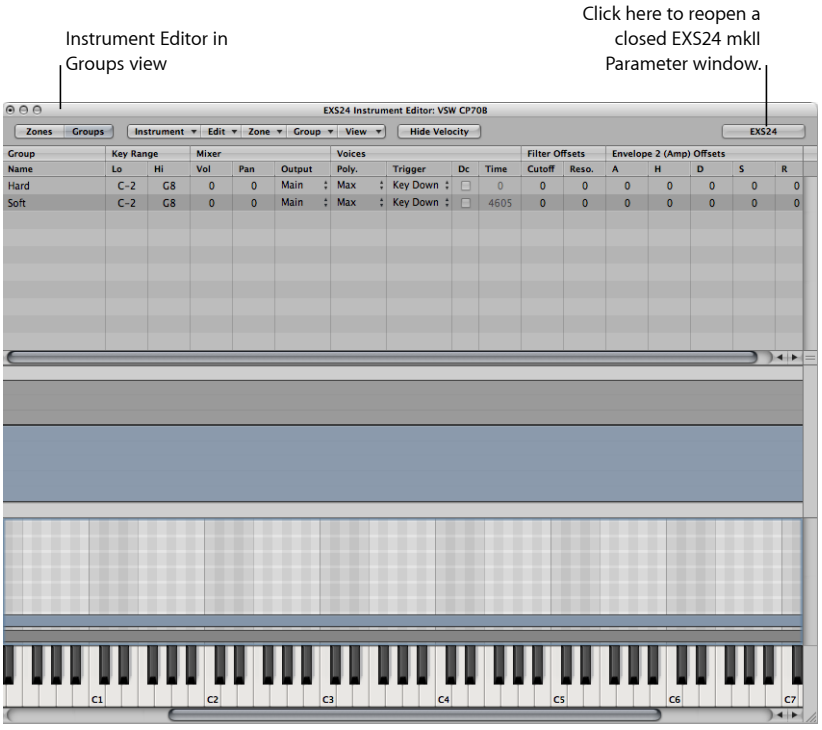


- *Zones column:* Displays all zones of the instrument. By default, every instrument contains All Zones (which includes "grouped" zones) and Ungrouped Zones icons. Click the desired icon to display the associated zones in the Parameters area.
- *Parameters area:* Displays the parameters of the zone (individual, all, or ungrouped) chosen in the Zones column.
- *Velocity area:* Shows the velocity range of the selected zone.

  Note: The Velocity area is turned off, by default.
- *Zones/Groups area:* Graphically indicates the zones or groups above the keyboard.
- *Keyboard:* Click notes to trigger the associated zone. The keyboard also serves as a visual reference for the placement of zones or groups (in the Zones/Groups area).

Getting to Know the EXS24 mkII Groups View Mode

In Groups view, the area above the keyboard displays Groups. The general menus, buttons, and so on, are displayed in both Zones and Groups views.



Note: Clicking the EXS24 button does not display the Parameter window in the foreground if it is covered by other floating windows.

Creating EXS24 mkII Instruments, Zones, and Groups

You can add zones and groups to loaded instruments, or you can create a new, empty instrument and fill it with zones and groups.

Important: The EXS24 mkII cannot directly record samples, as you would with a hardware sampler. You need to record the samples in a suitable application—Logic Express being the obvious choice.

To create a new instrument

- In the Parameter window (only when no sampler instrument is loaded), click the Edit button.
- In the Instrument Editor window, choose Instrument > New.

For information about loading sampler instruments, see Using the EXS24 mkII Sampler Instruments Pop-Up Menu. For information about saving, renaming and exporting sampler instruments, see Saving, Renaming, and Exporting EXS24 mkII Instruments.

For information about creating zones and groups, see the following sections:

Creating EXS24 mkII Zones

Quickly Creating Multiple Zones in the EXS24 mkII

Creating EXS24 mkII Groups

Creating EXS24 mkII Zones

A *zone* is a location into which a single sample—or audio file, if you prefer this term—can be loaded. The sample loaded into the zone is memory resident—it uses the RAM of your computer. A zone offers parameters that control sample playback. Each zone enables you to determine the range of notes that the sample spans—Key Range—and the root key—the note at which the sample sounds at its original pitch. In addition, sample start, end, and loop points, volume, and several other parameters can be adjusted for the zone. You can define as many zones as needed.

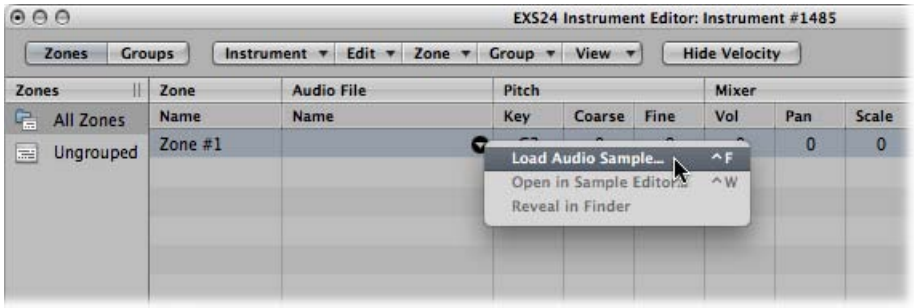
To first create a zone and then assign a sample to it

1. Choose Zone > New Zone (or use the New Zone key command).

   A new zone entry appears in the Instrument Editor.

2. Do one of the following:

   - Click the arrow in the Audio File column, then choose Load Audio Sample from the pop-up menu.



   - Double-click the empty area in the audio file column.

3. Choose the desired audio file in the file selector.

   - If you activate the "Hide used Audio Files" option, files used in the currently loaded EXS instrument are dimmed.

- The "Preview audio file in EXS instrument" option temporarily replaces the sample files in the currently selected zone. The zone is not directly triggered by activating this option, but it can be triggered by playing MIDI notes while the file selector is open—and different files are chosen. The selected sample can be heard as part of the zone, inclusive of all synthesizer processing (filters, modulation, and so on).

4 Click the Play button to loop playback of the currently selected sample file.

- Click the button a second time to stop playback.
- You can audition each file—in turn—by pressing the Play button, and then stepping through the files with the Down Arrow key, or by clicking them.

5 After you find a sample you'd like to use, click the Open button to add it to the zone. When the sample is loaded, the sample's name is displayed in the Audio File Name field.

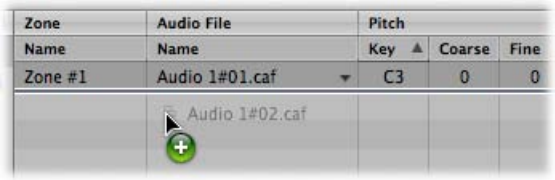
To create a zone by dragging an audio file to a key

- Drag an audio file onto one of the keys of the onscreen keyboard.

The start key, end key, and root key are all set to the note that the file was dropped on. This drag-and-drop functionality works for audio files from the following sources: Browser, Audio Bin, and the Finder.

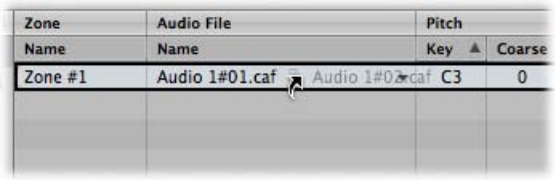
To create a zone by dragging an audio file to a range of keys

- Drag an audio file directly into the zone area to create a new zone.



The root key for the zone is the key at which the sample is played at its recorded pitch. This information is written in the sample header. If no root key is defined in the sample header, the C3 key is used by default.

Note: If you drag an audio file onto an existing zone, the file referenced by that zone is replaced with the new, dropped file. The cursor changes to reflect the replace mode.



Dragging a single sample into the empty area below Ungrouped Zones creates a default zone and a default group, with the new default zone placed in the default group.

Quickly Creating Multiple Zones in the EXS24 mkII

You can load multiple samples in one operation. The Instrument Editor automatically creates new zones and places the loaded samples into them.

To create multiple zones in one operation

1. Choose Zone > Load Multiple Samples in the Instrument Editor (or use the Load Multiple Samples key command).
2. Browse to the desired location, then use the Add or Add All buttons to select the samples you want to use.
3. Click the Done button when you are finished.
4. Choose one of the three automatic mapping modes in the Load Multiple Samples dialog:

- *"Auto map" by reading the root key from audio file:* Uses the *root keys* stored in the audio file headers, and places the samples, as zones, across the keyboard range. The number of keys that constitute a zone is intelligently determined by the placement of neighboring zones.
- *"Drums" zone without range, root key from audio file:* Uses the *root keys* stored in the audio file headers. Each zone is mapped to a single key on the keyboard—as determined by the root key information.
- *Contiguous zones:* Ignores all root key information and maps the samples across the keyboard in chromatic order. The Zone width field allows you to specify the width—the key range—of the newly generated zones. The Start Note field defines the start note of newly generated zones.

You can also load multiple samples by dragging them into the Instrument Editor. Dragging multiple samples into a group folder assigns the samples to the respective group. Dragging multiple samples below the Ungrouped Zones area assigns the audio files to a new default group.

Note: If you drag multiple files onto one of the keyboard keys, the Load Multiple Samples dialog does not include the Start Note field, as the start key, end key, and root key are all set to the note that the file was dropped on.

Creating EXS24 mkII Groups

Imagine that a drum kit has been created, with a number of different samples being used in several zones, mapped across the keyboard. In many musical circumstances, you might want to adjust the parameters of each sample independently—to alter the decay of the snare, or to use a different cutoff setting for the hi-hat samples, for example.

This scenario is where the EXS24 mkII's *groups* feature comes into play. Groups allow for very flexible sample organization. You can define as many groups as required and assign each zone to one of these groups. In a drum set, for example, you could assign all kick drums to Group 1, all snares to Group 2, all hi-hats to Group 3, and so on.

Why might you want to do this?

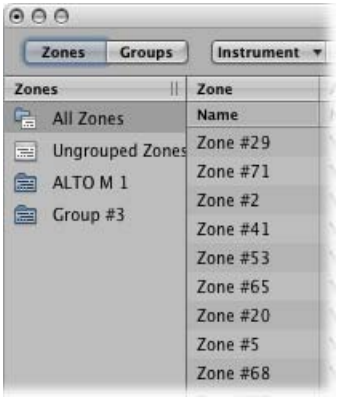
A group makes it possible to define a velocity range for all assigned zones, allowing you to specify a velocity window in which the grouped zones will be triggered, as one example. Each group also features offset parameters for the amplitude envelope and filter settings made in the Parameter window.

It's also possible to play all zones without defining and assigning any groups—in this situation, changes to parameter settings affect all samples in all zones equally.

To create a new group

- Choose Group > New Group in the Instrument Editor.

A new group appears in the Zones column at the left-hand side of the Instrument Editor.



To assign a zone to a group

Do one of the following:

- Select the group in the zone's Group menu.

- Select a zone in the EXS Instrument Editor, Finder, Audio Bin, or Browser—and drag it into a group displayed in the Zones column.
- Drag an ungrouped zone (or multiple selected zones) into the empty area below the Ungrouped Zones icon. This creates a new group, containing the dragged zone, or zones.
- Drag a zone (or multiple selected zones) out of one group:
  - into another group. This changes the previous group assignment to the new group.
  - onto the Ungrouped Zones icon. This changes the previous group assignment to unassigned (ungrouped).
  - to the empty area below the Ungrouped Zones icon. This creates a new group containing the dragged zone (or zones).

  Tip: Pressing Option while dragging zones to another group will copy, rather than move, the selected zones.

To delete all groups that do not have a zone assignment

- Choose Group > Delete Unused Groups in the Instrument Editor.

Editing EXS24 mkII Zones and Groups

Zones and groups offer unique parameters that enable you to customize your sampler instrument. You can use the zone parameters to edit the pitch, velocity range, panorama, looping parameters, and other aspects of zones. You can use the group parameters to adjust the velocity, output, and to offset envelopes and filters for a group of zones, as examples.

Editing techniques, menu selection commands, and other parameter interactions that are shared by zones and groups are discussed in these sections:

- Common EXS24 mkII Zone and Group Editing Commands
- Common EXS24 mkII Zone and Group Selection Commands
- Common EXS24 mkII Zone and Group Sorting Commands
- Showing and Hiding EXS24 mkII Zone and Group Parameters
- Graphical Editing of EXS24 mkII Zones and Groups

For information about parameters that differ between zones and groups, see Setting EXS24 mkII Zone Parameters and Setting EXS24 mkII Group Parameters.

Note: Click the EXS24 button in the top-right corner of the Instrument Editor window to reopen a closed Parameter window and bring it to the foreground. This button is dimmed when the Parameter window is open.

Common EXS24 mkII Zone and Group Editing Commands

Use the Edit menu for all basic sampler instrument editing operations, such as copying zones, undoing edit operations, and so on.

- *Undo:* Allows the most recent change to the sampler instrument to be undone.
- *Redo:* Undoes the last Undo command.
- *Cut, Copy, Paste:* The standard commands for cutting, copying, and pasting values. You can also cut, copy, and paste selected zones and groups.
  - When you copy groups in Zones view, the selected groups and their associated zones are copied. The group assignments of the zones are retained.
  - When you copy groups in Groups view, only the groups themselves are copied, not the associated zones.
- *Delete:* Deletes the currently selected zone or group.

Common EXS24 mkII Zone and Group Selection Commands

There are a number of ways that you can select zones and groups for editing.

The Edit menu contains the following zone and group selection commands:

- *Select All:* Selects all zones and groups of the loaded sampler instrument.
- *Toggle Selection:* Toggles the selection between the currently selected zones or groups and all currently unselected zones or groups.

You can also click zones and groups in the Parameters area:

- Clicking the parameters of a single zone or group selects that zone or group.
- Shift-clicking two nonadjacent zones in the Zones view selects these two zones, and all zones between the two clicked zones.
- Command-clicking multiple zones selects each clicked zone.
- You can use the Up Arrow key and Down Arrow key to select the previous and next zone or group.

You can also define a specific MIDI event for use as a group selection switch. See Using the EXS24 mkII Advanced Group Selection Parameters.

Switching EXS24 mkII Zones with a MIDI Keyboard: Select Zone of Last Played Key

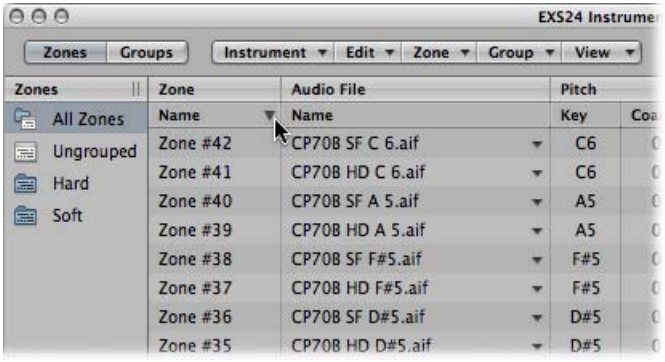
If you choose the “Select Zone of Last Played Key” command in the Zone menu, you can switch between zones by pressing a key on a connected MIDI keyboard. You can continue to select zones by clicking them in the editor when this feature is enabled.

Switching EXS24 mkII Groups with a MIDI Keyboard: Select Group of Last Played Key

If you choose the "Select Group of Last Played Key" command in the Group menu, you can switch between groups by pressing a key on a connected MIDI keyboard. This is useful when you want to adjust the velocity of an instrument's groups, for example.

Common EXS24 mkII Zone and Group Sorting Commands

You can easily sort zones and groups in the EXS24 mkII Instrument Editor by clicking the subcolumn heading that you wish to sort by. For example, if you want to sort your zones by name, click the Name subcolumn heading in the Zone column and your zones will be sorted alphabetically.



If you wanted to sort groups by lowest to highest starting velocity, for example, you would click the Low subcolumn heading in the Velocity Range column. The group with the lowest starting velocity range is displayed at the top of the list.

Click the triangle to invert—reverse—the sort order.

Showing and Hiding EXS24 mkII Zone and Group Parameters

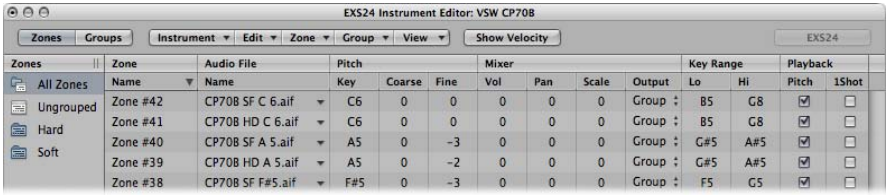
The View menu allows you to determine which zone and group parameters are shown in the Parameters area of the Instrument Editor:

- *View All:* Shows all available columns and subcolumns.
- *Individual Group and Zone display settings:* Choose the individual columns and subcolumns you want to display. The zone entries are available in Zone view. The group entries are available in Groups view.

  Tip: Press Option and choose a disabled zone or group column to limit the display to the chosen column.
- *Restore to Default:* Reverts to the default view.
- *Save as Default:* Saves the current view of zone and group parameters as the default view whenever you open the EXS24 mkII Instrument Editor.

Setting EXS24 mkII Zone Parameters

The zone parameters provide extensive control over each zone, or sample, in your sampler instrument.



- *Zone Name field:* Displays the zone name. New zones are automatically assigned a consecutive number. Click a zone number to enter a name.
- *Audio File menu:* Displays the audio file name. Move the pointer over a name to reveal a help tag with additional information, such as format, bit depth, sample rate, and so on. Press Command before the help tag appears, to display the full file path in the help tag. Click the arrow to open a shortcut menu that offers the following commands:
  - *Load Audio Sample:* Opens a file selection box, allowing you to select an audio file. Default key command: Control-F.
  - *Open in Sample Editor:* Opens the selected sample in the Logic Express Sample Editor (or the sample editor chosen in the Open External Sample Editor preference). Default key command: Control-W.
  - *Reveal in Finder:* Shows the full path of the loaded audio file in the Finder.

  Tip: Double-click the name of a sample in the Audio File > Name column to open the audio file in the Sample Editor. When no audio file is loaded, the audio file selector opens.
- *Pitch fields:* Key determines the root note of the sample—in other words, the note at which the sample will sound at its original pitch.
  - Use the Coarse and Fine fields to tune the sample in semitone/cent increments.
- *Volume field:* Adjusts the overall output level of the zone.
- *Pan field:* Adjusts the pan position of the zone. This parameter works only when the EXS24 mkII is used in stereo.
- *Scale field:* Balances the level of a sample across the selected key range. A negative value makes notes lower than the root key position louder than higher notes; positive values have the opposite effect.
- *Output menu:* Determines the outputs used by the zone. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through 16. This allows individual zones to be routed independently to aux channel strips (in a multi-output EXS24 mkII instance).

- *Key Range fields:* The two key range parameters allow you to define a key range for the zone.
  - *Lo(w):* Sets the lowest note for the zone.
  - *Hi(gh):* Sets the highest note for the zone.

  Playing notes outside this range will not trigger the sample assigned to this zone.
- *Pitch checkbox:* Activate to change the sample pitch when triggered by different keys. When disabled, the sample is always played at its original pitch, regardless of the note played.
- *1Shot checkbox:* Causes the zone to ignore the length of incoming MIDI note events—resulting in the sample always being played from beginning to end whenever a note-on event is received. This is useful for drum samples, where you often don't want the MIDI note length to affect sample playback. Also see the Fade field parameter below.
- *Reverse checkbox:* Enable to play the sample from the end to the beginning.
- *Group menu:* Shows the group assignment of a zone. For more information, see Creating EXS24 mkII Groups and Setting EXS24 mkII Group Parameters.
- *Velocity Range checkbox and fields:* Activate the checkbox to define a velocity range for the zone.
  - *L(ow):* Sets the lowest velocity that will trigger the zone.
  - *Hi(gh):* Sets the highest velocity that will trigger the zone.

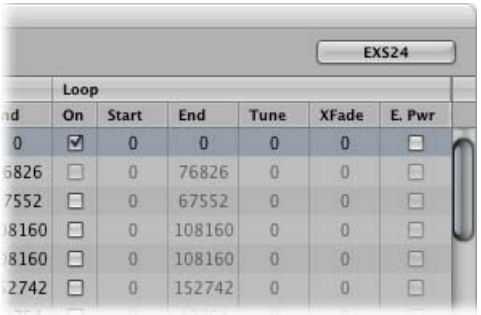
  Playing notes outside this velocity range will not trigger the sample assigned to this zone.
- *Sample Start and End fields:* Set the sample start and end points, respectively. Control-click either field to open a menu that allows you to open the sample in the Logic Express Sample Editor (or an external editor), where you can set the start and end points graphically. See Editing Samples in the EXS24 mkII Sample Editor.
- *Fade field:* Determines the fade out time for a one-shot sample. The value is shown in samples. The difference between the value specified in this field and the value shown in the End field determines the fade out duration. The lower the value, the longer it takes for the sample to reach a 0 level (at the sample end point).

  This option is dimmed when the Loop On checkbox is active. See Using the EXS24 mkII Zone Loop Parameters for details about zone loop functions.

  Note: This parameter defaults to a value of 0, except in cases when the Sampler Instrument is created with Logic Express's Audio > "Convert Regions to New Sampler Track" command. This function uses transient markers, and results in a default Fade field value that matches the slicing offset of the following transient marker.

Using the EXS24 mkII Zone Loop Parameters

The EXS24 mkII can loop playback of either an entire sample or a portion of it, when sustained MIDI notes are received.



- *Loop On checkbox:* Activate to enable looping and to allow access to the other Loop parameters.
- *Loop Start, Loop End fields:* Define discrete loop start and end points, allowing you to loop a portion of the audio file.
  - Control-click either field to open a shortcut menu that allows access to the Logic Express Sample Editor (or an external editor). This enables you to set the loop start and end points graphically: Loop Start is represented by the LS marker and Loop End by the LE marker. See Editing Samples in the EXS24 mkII Sample Editor.
- *Tune field:* Changes the tuning of the looped portion of the audio file in cent increments.
- *Xfade (Crossfade) field:* Determines the crossfade time between the end/start of a looped sample. In a crossfaded loop, there is no "step" between the loop end and loop start points. The higher the value, the longer the crossfade and the smoother the transition between the loop end and start points. This is especially convenient with samples that are hard to loop and that would normally produce clicks at the transition point—the join in the loop.
- *E. Pwr (Equal Power) checkbox:* Activate to enable an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This will fade out/fade in the join between the loop end/start at an equal volume level.

Note: The ideal settings for the Xfade and E. Pwr parameters depend on the sample material. A loop that cycles reasonably smoothly is the best starting point for a perfectly crossfaded loop, but a crossfaded loop does not always sound better. Experiment with both parameters to learn how, when, and where they work best.

Setting EXS24 mkII Group Parameters

Group parameters provide simultaneous control of all assigned zones.

- *Group Name field:* Displays the group name. Click to enter a name.
- *Key Range fields:* Define a key range for the group.
  - *Lo(w):*Sets the lowest note for the group.
  - *Hi(gh):*Sets the highest note for the group. Playing notes outside this range will not trigger the zones assigned to this group.

  Note: Take your time with these parameters, as they override zone range settings, possibly making some zones inaudible.
- *Vol(ume):* Adjusts the overall level of the group—and, therefore, the volume of all zones in the group. This works much like a subgroup on a mixing console.
- *Pan:* Adjusts the pan position of the group—stereo balance for stereo samples—and the pan position of all assigned zones simultaneously.

  Note: This will affect any individual zone panning adjustments.
- *Output:* Determines the outputs used by the group. Choices include the main outputs, and paired channels 3 and 4, 5 and 6, 7 and 8, 9 and 10, or individual outputs 11 through 16. This allows individual groups to be routed independently to aux channels in a multi-output EXS24 mkII instance.

  Note: This will have an impact on any individual zone output assignments.
- *Poly. (Polyphony):* Determines the number of voices that the group can play. The Max option ensures that the group uses all voices allowed by the Voices parameter in the Parameter window. For an example of how this parameter can be used, see Hi Hat Mode: Using the EXS24 mkII Poly Parameter.
- *Trigger menu:* Determines if zones pointing to this group are triggered on key down (Key Down setting) or on key release (Key Release setting). This is useful for emulating organ key clicks, for example, where you may want the organ note triggered on key down, but the organ click triggered on key release.
- *Dc (Decay) checkbox and Decay Time field:* Select the checkbox to access the Decay Time parameter.
  - *(Decay) Time field:*Determines the time it takes for the level of a sample (triggered by key release) to decay.

Note: The Decay parameters will function only when the Trigger parameter is set to Key Release.

- *Cutoff and Reso(nance) fields:* Independently offsets the Cutoff and Resonance settings for each group. This can be useful if you want the initial impact of a note to be unfiltered for one group but not others.
- *Envelope 1/Envelope 2 Offsets fields:* Independently offsets the envelope settings in the Parameter window for each group. This is useful if you want the filter (Envelope 1) or volume (Envelope 2) envelopes to affect the samples in a group—after the initial impact of the triggered sounds.
  - *H (Hold) field:* Determines the time that the envelope will be held at the maximum attack level, before the decay phase begins.

  Note: When the Trigger parameter is set to Key Release, the Decay Time parameter controls the decay level, rather than Envelope 2 (the volume envelope). This means that when Trigger is set to Key Release, the Envelope 2 Offsets have no effect.
- *Vel(ocity) Range:* Sets a velocity range for the group. Velocity Range is useful for sounds where you want to dynamically mix—or switch between—samples, by playing your MIDI keyboard harder or softer. This feature is ideal for layered sounds, such as a piano/string layer, or when switching between different percussion samples, for example.
  - *Lo(w):* Sets the lowest velocity that will trigger the group.
  - *Hi(gh):* Sets the highest velocity that will trigger the group. Playing notes outside this velocity range will not trigger the zones assigned to this group.

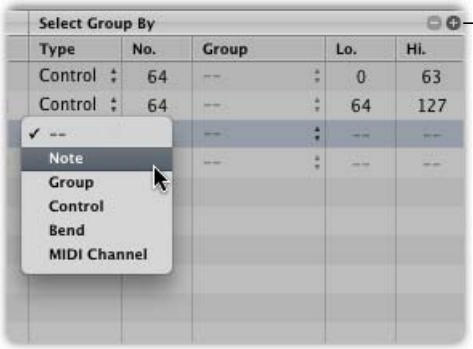
  Note: The settings made here override zone settings: When a zone's velocity range is larger than the group setting, the zone's velocity range is limited by the group setting.

Hi Hat Mode: Using the EXS24 mkII Poly Parameter

A practical use of the Poly parameter is to set up a classic "hi-hat mode" within a full drum kit that is mapped across the keyboard. In this scenario, you could assign an open and closed hi-hat sample to a group, and set the Voices parameter of the group to 1. In this example, the most recently triggered of the two hi-hat samples will mute the other, as only one voice is allowed for the group. This mirrors the real-world behavior of hi-hats. The other sounds of the drum kit can still be played polyphonically, if samples in zones are assigned to another group.

Using the EXS24 mkII Advanced Group Selection Parameters

You can define a specific MIDI event for use as a group selection switch. Whenever the defined selection event is triggered, zones pointing to this group can be played, and other groups (selected with a different event) are not played. The defined event does not play or alter a sound; it simply acts as a group selection switch.



This feature works with MIDI note, controller, and pitch bend events (on a specified MIDI channel). You first need to define a group number as the "base" for the Select By command. After the group number is defined, selecting a particular group means that only zones pointing to that group will play, and other groups are not played.

For example, if you want the EXS24 mkII to automatically switch between two string sample groups—one for staccato samples and one for legato samples—you could set the Select Group By menu to MIDI notes, and assign a different MIDI note to trigger each group. This way, you can use a note (that is not triggering a sound) as a remote group switch.

You can further refine the group selection conditions by clicking the plus sign in the upper-right corner of the Select Group By column. To continue the staccato and legato string layer example, you could specify that a controller message switches between different articulations. For another example of how this feature can be used, see Remapping of Pitch Bend and Modulation Wheel Events in the EXS24 mkII.

Click the minus sign to remove a Select Group By condition, and broaden the group selection criteria.

Remapping of Pitch Bend and Modulation Wheel Events in the EXS24 mkII

In order to create realistic-sounding performances in an easy and intuitive way, the Jam Pack 4 (Symphony Orchestra) instruments use the modulation wheel to switch between articulations—legato, staccato, and so on. The pitch bend wheel is used to change expression—crescendo, diminuendo, and so on. Further information about this can be found in the Jam Pack 4 documentation.

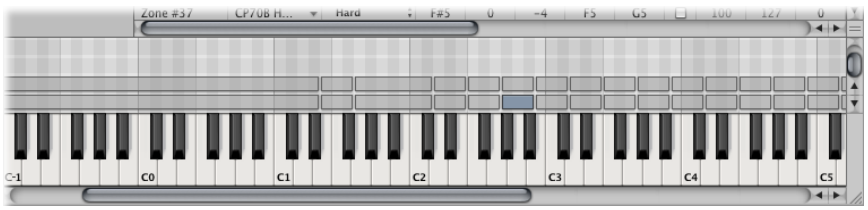
This is achieved by internally remapping pitch bend events to MIDI controller 11 and modulation wheel events to MIDI controller 4. To ensure compatibility with the Jam Pack 4 instruments, the EXS24 mkII automatically uses this remapping behavior for Jam Pack 4 instruments.

You can also use this remapping model for other instruments by choosing the “Map Mod & Pitch Wheel to Ctrl 4 & 11” item in the Instrument menu.

The EXS24 mkII will remap incoming pitch bend and mod wheel events to controller 11 or controller 4, respectively. The default pitch bend and modulation wheel functionality cannot be used when in this mode.

Graphical Editing of EXS24 mkII Zones and Groups

You are not limited to editing zones and groups in the Parameters area. You can also graphically edit a number of zone and group parameters in the Zones or Groups Display area above the keyboard. If you want to edit the audio file of a zone, see Editing Samples in the EXS24 mkII Sample Editor.

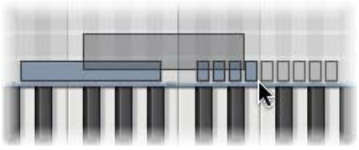


To move a zone or group

- Drag the zone or group to the desired position.

To move multiple zones or groups

- Shift-click or drag to select the zones or groups, and drag them to the desired position.



To change the root key when moving a zone

- Hold down Command-Option while dragging the zone.

To change the start or end note of a zone or group

1. Move the cursor to the beginning or end of a zone or group (the cursor will change to the resize icon).

2. Drag the start or end point of the zone or group to the desired position.

To move a zone to the left or right

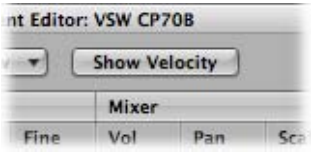
- Use one of the following key commands:
  - *Shift Selected Zone(s)/Group(s) Left:* Option–Left Arrow key
  - *Shift Selected Zone(s)/Group(s) Right:* Option–Right Arrow key

To shift both the root note and the zone position

- Use one of the following key commands:
  - *Shift Selected Zone(s)/Group(s) Left (Zones incl. Root Key):* Shift–Option–Left Arrow key
  - *Shift Selected Zone(s)/Group(s) Right (Zones incl. Root Key:* Shift–Option–Right Arrow key

To edit the velocity range of a zone or group

1. Click the Show Velocity button at the top of the Instrument Editor (or use the Show/Hide Velocity key command).



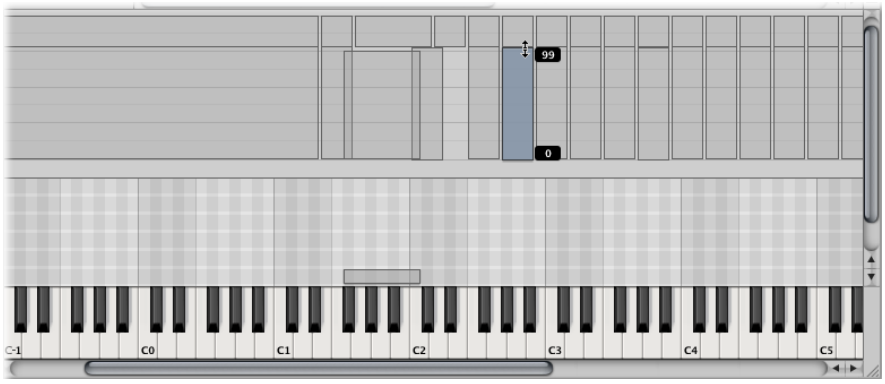
The Velocity Display area opens above the Zones or Views Display area.

2. Click one or more zones or groups in the Display area.

The velocity bars of the selected zones/groups is highlighted in the Velocity Display area.

3 Move the cursor to either the High or Low value of the velocity bar that you want to change (the cursor will change to the resize icon).

4 Drag upward to raise the value, or downward to lower the value.



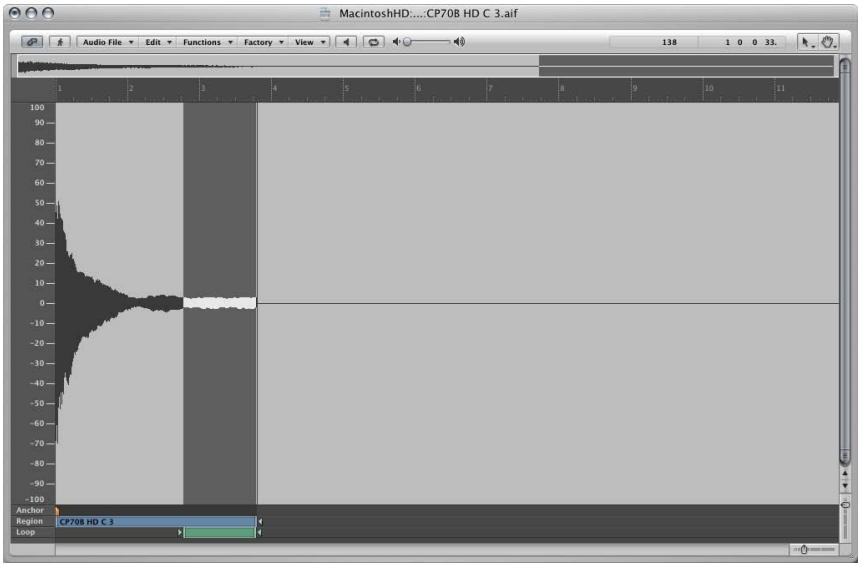
Saving, Renaming, and Exporting EXS24 mkII Instruments

You can access all basic sampler instrument file operations in the Instrument Editor's Instrument menu.

- *Save:* Saves the currently loaded sampler instrument. When you create a new instrument and save it for the first time, you will be asked to provide a name. If you have edited an existing sampler instrument and use this command, the existing file name is used and the original instrument is overwritten. You can also use the Save Instrument key command.
- *Save As:* Saves the currently loaded sampler instrument, but you will be prompted to provide a (different) file name. Use this command when you want to save a copy, or multiple versions of, an edited sampler instrument, rather than overwriting the original version.
- *Rename:* Renames the loaded sampler instrument. The renamed version replaces the original version on the hard disk.
- *Export Sampler Instrument and Sample Files:* Copies the selected sampler instrument (including all associated audio files) to another folder location. Choosing this command opens a standard file selector. You may browse to an existing folder or create a new folder. You can also use the Export Sampler Instrument and Sample Files key command (default: Control-C).

Editing Samples in the EXS24 mkII Sample Editor

Because the EXS24 mkII and Logic Express Sample Editor are built to work together, the EXS24 mkII doesn't require a built-in graphical editor. The most intuitive way to adjust sample and loop start and end points is by working directly on a visual representation of the waveform.



To open the Sample Editor

- Control-click either the Loop Start or Loop End parameter fields of the zone you want to edit in the Instrument Editor window.

  This opens a shortcut menu, which you can use to open the selected sample in the Logic Express Sample Editor (or the external sample editor set in the preferences—see Using an External Instrument Editor with the EXS24 mkII).

To edit sample borders and loop points

1. After the sample is opened in the Sample Editor, you can drag the sample borders and loop points graphically. Loop Start is represented by the LS marker. Loop End is represented by the LE marker.

   The Sample Editor has a number of convenient loop commands that may accelerate your workflow. See Working with the EXS24 mkII Sample Editor Loop Commands.

2. When you have completed your edits, save the sample. The new values written to the audio file header are used by the EXS24 mkII.

   Note: Edited samples may have values that are not accurately shown in the Instrument Editor. See Updating Edited EXS24 mkII Zone Information.

Updating Edited EXS24 mkII Zone Information

After you have saved and reopened a sample that was edited in either the Logic Express Sample Editor or a sample editor not made by Apple, it is likely that either the start and end, or loop point values—shown in the Parameters area—will no longer be accurate.

To update zone information

- Open the Zone menu and choose the "Update Selected Zone(s) Info from Audio File" command.

This feature reads loop settings and start and end points directly from the audio file, and updates the settings of the zone—shown in the Parameters area—accordingly.

Working with the EXS24 mkII Sample Editor Loop Commands

The Edit menu of the Sample Editor includes the Sample Loop → Selection, Selection → Sample Loop, and "Write Sample Loop to Audio File" items.

To use the Sample Editor Loop commands

1. Choose either of the selection commands in the Edit menu of the Sample Editor:
   - *Sample Loop → Selection:* The loop area—defined by the Loop Start and End points—is used to select a portion of the overall audio file.
   - *Selection → Sample Loop:* The selected area is used to set the Loop Start and End points.
2. After you select the desired area using either of these commands, choose Edit > "Write Sample Loop to Audio File."

The new loop values are written to the audio file header.

Using an External Instrument Editor with the EXS24 mkII

The EXS24 mkII allows you to use external applications such as Redmatica's Keymap to edit your EXS instruments.

To open your EXS instruments in an external instrument editor

1. Choose Options > "Open in external Instrument Editor" in the EXS24 mkII Parameter window.
2. In the file selector, locate and select the desired instrument editor software.
3. Repeat step 1, but choose the "Open in [name of external instrument editor]" command. This is the same command as above, but it is automatically renamed after an external instrument editor is assigned.
4. Edit the instrument in the external instrument editor, then use the external instrument editor to send the instrument back to Logic Express.

Important: You need to save the edited instrument in Logic Express—not the external editor—to make any instrument changes permanent.

To reassign the external instrument editor

- Hold down Option while choosing Options > "Open in [name of external instrument editor]."

Importing EXS24 mkII Sampler Instruments

The EXS24 mkII is compatible with the AKAI S1000 and S3000, SoundFont2, SampleCell, DLS, Gigasampler, and ReCycle sample formats, as well as the Vienna Library. See these sections:

Importing SoundFont2, SampleCell, DLS, and Gigasampler Files

Converting ReCycle Files to EXS Instruments

Converting AKAI Files with the EXS24 mkII

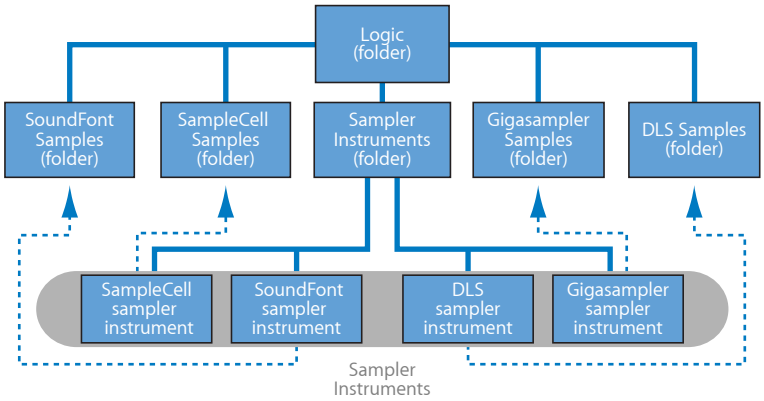
Importing SoundFont2, SampleCell, DLS, and Gigasampler Files

The EXS24 mkII recognizes SoundFont2, SampleCell, DLS, and Gigasampler files placed inside the Sampler Instruments folder, and converts them into sampler instruments.

To import SoundFont2, SampleCell, DLS, or Gigasampler files into the EXS24 mkII

1. Copy or move your SoundFont2, SampleCell, DLS, or Gigasampler files into the ~/Library/Application Support/Logic/Sampler Instruments folder.
2. Choose the SoundFont2, SampleCell, DLS, or Gigasampler file in the EXS24 mkII Sampler Instruments pop-up menu. The EXS24 mkII automatically converts the chosen file into an EXS sampler instrument:
   - An EXS instrument file is created in the Sampler Instruments folder. This contains the file in its original format.
   - The raw samples associated with the sampler instrument are placed in one of the following folders, depending on the format being converted:
     - ~/Library/Application Support/Logic/SoundFont Samples
     - ~/Library/Application Support/Logic/SampleCell Samples
     - ~/Library/Application Support/Logic/Gigasampler Samples

- ~/Library/Application Support/Logic/DLS Samples



The procedure outlined above also applies when you are importing SoundFont2 and SampleCell *Bank* files. These file types contain multiple sounds, in addition to single-instrument files.

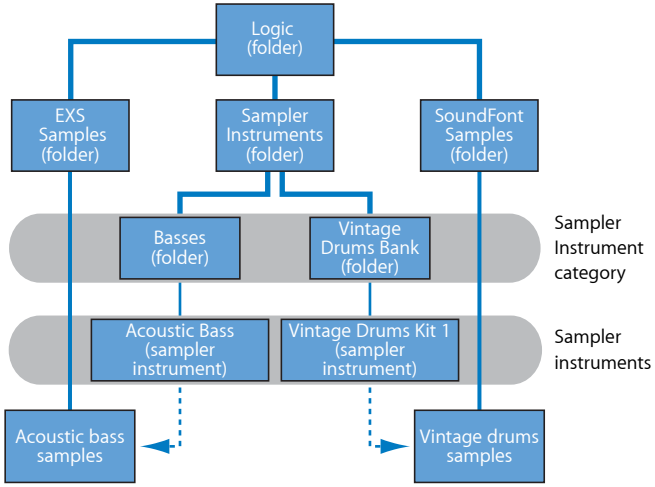
When you load a SoundFont2 or SampleCell Bank into the EXS24 mkII, it creates a Bank and a Samples folder, named after the SoundFont2/SampleCell Bank file. The word "Bank" or "Samples" is appended to each folder name.

An EXS sampler instrument file is automatically created for all sounds in the bank, and placed in the new *Bank* folder. The Sampler Instruments pop-up menu automatically updates to reflect the new folder hierarchy. All samples associated with the bank are automatically added to a Samples folder inside the SoundFont/SampleCell folder.

For example, if you load a SoundFont2 bank file named "Vintage Drums," which contains over 50 individual drum kits from several different vintage drum machines, into the EXS24 mkII, the following will happen:

- A new folder named "Vintage Drums Bank" will be created in the ~/Library/Application Support/Logic/Sampler Instruments folder.
- A second folder named "Vintage Drums Samples" will be created in the ~/Library/Application Support/Logic/SoundFont Samples folder.

- The Sampler Instruments pop-up menu hierarchy is updated and the original Vintage Drums entry is replaced with a “Vintage Drums.Bank” entry. This new entry is a folder that contains the individual sampler instruments, which can be selected and loaded as usual.



After conversion is complete, the original SoundFont2, SampleCell, or Gigasampler source files can be freely deleted from the hard disks.

Note: You can store your imported sampler instruments in any folder on any of your computer’s hard drives. To access these instruments from the Sampler Instruments pop-up menu, you need to create an alias pointing to the target folder within the ~/Library/Application Support/Logic/Sampler Instruments folder.

Converting ReCycle Files to EXS Instruments

ReCycle, a sample editing program from Propellerhead Software, can generate a number of file types that can be read by Logic Express and the EXS24 mkII.

ReCycle separates sample material into small segments called *slices*, based on waveform peaks, or transients, in the audio file. This way, ReCycle is able to split an audio file into musically relevant slices. The timing of these slices—in a drum loop, for example—is also mapped to an automatically generated region, in Logic Express.

The EXS24 mkII supports the following ReCycle file types:

- *Old ReCycle file:* These files have the *.rcy* suffix. The abbreviation for this file type is *RCSO*. These files are not commonly used these days.
- *Old ReCycle export file:* These files have the *.rex* suffix. The abbreviation for this file type is *REX*. A number of older sample libraries include REX format files.

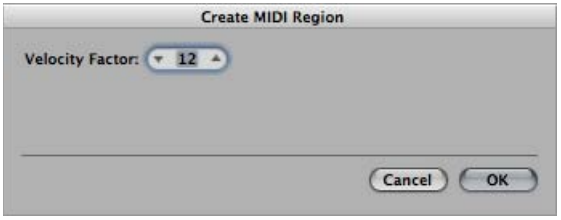
- *ReCycle 2.0 file:* These files have the *.rx2* suffix. The abbreviation for this file type is *REX2*. These files are used extensively by Propellerhead Reason, and many popular sample libraries include REX2 format files.

Generating a Zone for Each Slice

The "Extract MIDI Region and Make New Instrument" command creates a new EXS24 instrument from a ReCycle file, and generates an independent zone for each slice.

To create a new EXS instrument and assign each slice to a zone

1. Choose Instrument > ReCycle Convert > "Extract MIDI Region and Make New Instrument" in the Instrument Editor.
2. Choose the desired ReCycle file in the file selector, then click Open.
3. Enter a velocity factor in the Create MIDI Region window (see below for region-related information).



The velocity factor analyzes the loudness—transient peak—of each slice in the imported ReCycle file. It then maps this value to a corresponding velocity value for the MIDI note event that is used to trigger the slice.

- If a positive value (up to 100) is entered, louder slices will generate MIDI note events with higher velocity values.
- The use of negative values on louder slices will result in the generation of lower MIDI velocity values for each note event.

4. Click OK.

The EXS24 mkII generates a zone for each slice of the imported ReCycle file and assigns these zones to a single group. The new EXS instrument is named after the ReCycle loop. If an EXS instrument of that name already exists, a number sign (#) and a number will be appended to the name. For example, if you import a ReCycle file named "Tricky Backbeat" but "Tricky Backbeat" already exists as a sampler instrument, the imported instrument would be named "Tricky Backbeat#1," thus ensuring that the filename is unique within the Sampler Instruments folder.

In addition, a MIDI region is generated on the currently selected track, at the current project position, rounded to whole bars. This MIDI region is used to trigger the imported slices at the timing defined by the ReCycle file. You can generate new MIDI regions at any time from the imported EXS instrument (see Generating a MIDI Region from a ReCycle Instrument), so feel free to modify or delete the region.

The "Extract MIDI Region and Add Samples to Current Instrument" command allows you to add the slices of a ReCycle loop to any EXS instrument currently opened in the Instrument Editor. This allows you to use several different ReCycle loops in a single sampler instrument.

Assigning the Complete ReCycle Loop to a Zone

The Instrument > ReCycle Convert > "Slice Loop and Make New Instrument" command creates an EXS instrument from a ReCycle loop. Each slice is chromatically mapped across the keyboard—from low to high notes. Each zone plays back the ReCycle loop to the end point, at the current project tempo. This means that the lowest zone will play back the entire loop, but the highest zone will only play the last slice of the loop. Notes between the lowest and highest zones will play several slices.

This allows for old school drum'n'bass-style note triggering, where the sample loop start point is determined by playing the respective notes on the keyboard.

The Instrument > ReCycle Convert > "Slice Loop and Add Samples to Current Instrument" command adds the zones from the sliced loop to the currently active sampler instrument.

Pasting Loops from the Clipboard

The Edit > "Paste ReCycle Loop as New Instrument" command creates an EXS instrument from a ReCycle loop that was copied into the Clipboard—with ReCycle's Copy Loop function.

Instrument creation is identical to the "Extract MIDI Region and Make New Instrument" command.

The Edit > "Paste ReCycle Loop to Current Instrument" command adds the zones to the currently active sampler instrument.

Generating a MIDI Region from a ReCycle Instrument

You can generate a MIDI region from imported ReCycle files. These regions trigger the imported slices at the timing defined by the ReCycle files.

To generate a new MIDI region from a ReCycle instrument

- Choose Instrument > ReCycle Convert > "Extract Region(s) from ReCycle Instrument."

MIDI regions are created on the currently selected track, at the current project position, rounded to bars. A single MIDI region is generated for each imported ReCycle loop in the currently open instrument. This function will also ask for a velocity factor (see Generating a Zone for Each Slice).

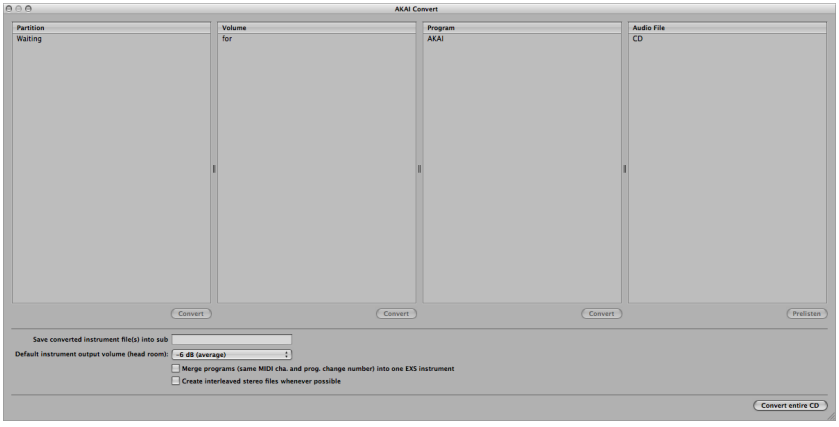
Converting AKAI Files with the EXS24 mkII

The EXS24 mkII can import samples in the AKAI S1000 and S3000 sample formats. The AKAI Convert feature can be used to import:

- An entire AKAI format CD-ROM disc
- An AKAI partition
- An AKAI volume
- An AKAI program
- An individual audio file (sample)

To convert AKAI files

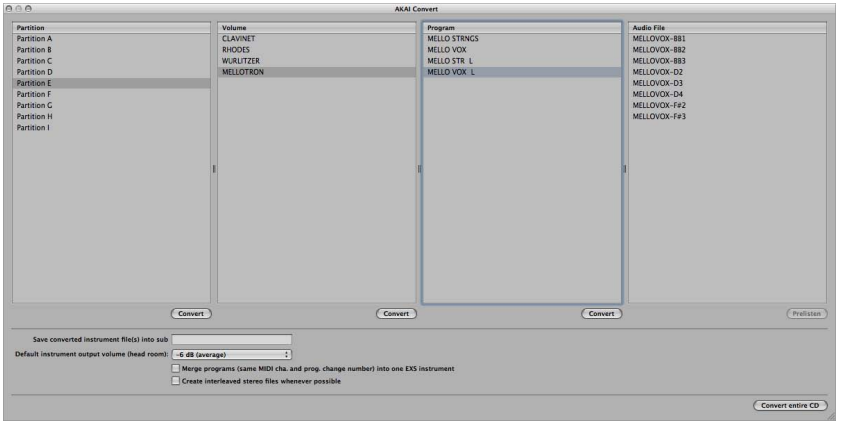
1 Open the Options pop-up menu in the Parameter window and choose AKAI Convert. This opens the AKAI Convert window, with "Waiting for AKAI CD" displayed across the four columns. See EXS24 mkII AKAI Convert Window Parameters for more information.



2 Insert an AKAI format sample disc into your optical drive. The display updates to show the contents of the disc. The Partition column will display information—with Partition A, Partition B, and so on, entries listed.

3 To view the contents of the partitions, click the desired entry to display the volume information contained within the partition.

4 To navigate deeper into the folder hierarchy of the disc, click the Volume entries to view any programs contained therein. Click the Program entries to view the raw audio files, or samples.



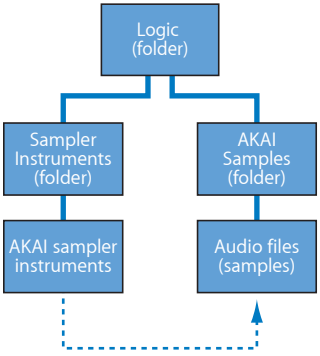
You can use the Prelisten button below the Audio File column to individually audition AKAI audio files before deciding whether to import them.

5 If desired, set any of the additional AKAI Convert parameters at the bottom of the window (see EXS24 mkII AKAI Convert Window Parameters).

6 After you have chosen a partition, volume, or program, click the Convert button beneath the corresponding column.

Note: If you want to convert an entire AKAI CD-ROM disc, click the "Convert entire CD" button in the lower-right corner of the AKAI Convert window.

The selected partition, volume, or program will be imported, along with all associated audio files.



- Imported audio files are stored in a folder named after the volume. This folder is created within the ~/Library/Application Support/Logic/AKAI Samples folder.

- Sampler instruments created by the import process match the program names. They are placed inside the ~/Library/Application Support/Logic/Sampler Instruments folder, or the sub-folder determined by the "Save converted instrument file(s) into sub" parameter. Sub-folders (named after the volume) are created when you convert a partition. If a volume contains only a single program, no sub-folder is created. Sub-folders named after the partition are created when you convert more than one partition.

The Sampler Instruments pop-up menu displays converted instruments, as shown:

EXS24 mkII AKAI Convert Window Parameters

The following section explains the parameters contained in the EXS24 mkII AKAI Convert window. For more information on converting AKAI files to EXS instruments, see Converting AKAI Files with the EXS24 mkII.

- *Save converted instrument file(s) into sub:* Use this when importing an entire disc. This creates a folder name that reflects the disc's name. Alternatively, you may want to save your converted instruments into a sub-folder based on a category, such as Strings. Then, if your AKAI disc consists of string samples, any imported programs or volumes will be added to the Strings sub-folder.
  - To enter a name for a sub-folder, click in this parameter field, enter the desired name, and press Return. All imported volumes and programs will automatically be added to this folder.

  Note: If an existing name is used, the imported sampler instrument will be *added* to the folder. It will *not* create a new folder of that name.

- *Default instrument output volume (head room):* Nondestructively alters the main Output volume in the Parameter window. This can be adjusted after conversion. Sustained pad sounds and polyphonic instruments in AKAI format often tend to have a higher output than a drum groove, for example. This can result in the output levels of some converted AKAI instruments being much higher than the rest of your EXS24 mkII sampler instrument library—occasionally, converted programs may be so loud that they clip. Set this parameter to the desired amount, which will limit the headroom (output level) of the EXS24 mkII parameter settings for each converted AKAI program. The value to choose for a given AKAI disc is a matter of trial and error, but here are some suggestions:
  - For drum discs, start with either no change (0 dB) or a headroom value of –3 dB.
  - For piano, string, or pad discs, you might try a headroom value of –9 dB.
  - For very loud programs, such as layered analog synthesizer instruments, you might even want to try –12 dB.
  - In cases where you're not sure of which headroom value to select, start with the average setting of –6 dB.
- *Merge programs (same MIDI cha. and prog. change number) into one EXS instrument:* Many discs created for AKAI samplers feature programs that contain single velocity layers for an instrument. AKAI samplers require that an entire volume, or all necessary single programs, are loaded in order to play all velocity layers. All of these single programs are automatically assigned to the same MIDI channel, and they also respond to the same MIDI program change number. The EXS24 mkII AKAI conversion intelligently checks for these settings and builds a single EXS sampler instrument out of multiple single programs. In general, when you are importing samples of this type, this option should be switched on.
  - The same is true for drum discs where single programs contain one instrument from a complete drum kit—kick, snare, hi-hat, and so on, as separate entities. You'll probably want these single AKAI programs to be merged into a single EXS sampler instrument—as a full drum kit.
  - There are, however, a number of AKAI discs where a single program of an AKAI volume contains the entire instrument, and where other programs in the same volume have the same MIDI channel and MIDI program change number preset. On this type of disc, use of the merge programs parameter is not desirable, and the option should be switched off.
- *Create interleaved stereo files whenever possible:* Should always be left enabled, as interleaved files offer better performance within the EXS24 mkII. When you are converting AKAI format samples, some audio files are created as split stereo *and* as interleaved stereo files. The detection of when it is possible to build an interleaved file is based on information stored with both the AKAI program and audio files. Both the left and right files must have the same settings; otherwise they cannot be used to create an interleaved file.

Managing EXS24 Sampler Instruments

As your sample library grows, the list of sampler instruments will also expand. To help you keep the list of sampler instruments manageable, the EXS24 mkII features a simple but flexible file management method.

To organize your sampler instruments into a preferred hierarchy

1. Create a folder in the Finder—Basses, for example—and drag it into the desired Sampler Instruments folder.
2. Drag the desired EXS24 mkII sampler instruments into this newly created folder.

The modified menu structure will be reflected when you open the Sampler Instruments pop-up menu.

Note: You will need to choose the Refresh Menu command in the Sampler Instruments pop-up menu after changes are made to the folder hierarchy in the Sampler Instruments folder.

The Sampler Instruments pop-up menu will display submenus only for folders that actually contain EXS instrument files. Other folders are not added to the menu. Aliases pointing to folders that contain EXS instrument files outside the Sampler Instruments folders can also be added to the menu. Even the Sampler Instruments folder itself can be an alias to a folder on a different drive or in another location.

For information on copying sampler instruments to your hard drive, and backing up sampler instruments used in a project, see Copying EXS24 Sampler Instruments to Your Hard Drive and Backing Up EXS24 Sampler Instruments.

Copying EXS24 Sampler Instruments to Your Hard Drive

It is strongly recommend that you copy any EXS sampler instruments, along with all associated audio files, to your hard drive. This way, you always have direct, immediate access to your sampler instruments without searching for and inserting CD-ROM or DVD discs. This also enables you to organize your sampler instruments to meet your needs. In addition, load times are faster, and you can play back samples that exceed the size of your computer's RAM by "streaming" them from your hard disk. This feature is not suitable for optical drives.

To copy sampler instruments to your hard drive

1. Copy the sampler instrument file into the ~/Library/Application Support/Logic/Sampler Instruments folder.
2. Copy the associated samples into a folder named Samples in the same folder as the Sampler Instruments folder.

Backing Up EXS24 Sampler Instruments

You can use the "Backup audio files of all used and active instruments of current project" key command to copy the audio and sampler instrument files (of all currently active sampler instruments in the project) to a specified file location. Folders for the audio files associated with these sampler instruments are created in the target location.

This makes it easy to keep all of your sampler instruments and audio samples in one place, and ensures that every project folder contains all the necessary sampler instruments and audio files you will need, even if you don't have access to your full sampler instrument library.

Tip: You can also do this by setting up your project to copy EXS24 sampler instruments and samples into the project folder. For more information, see the *Logic Express User Manual*.

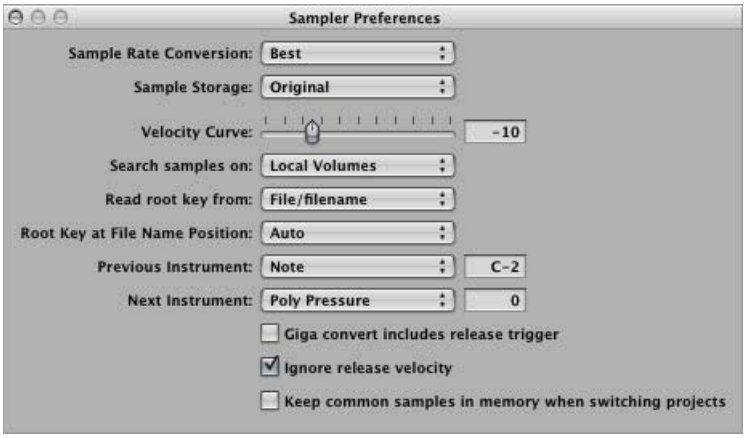
Setting EXS24 mkII Sampler Preferences

The EXS24 mkII Sampler Preferences window provides access to sample-related preferences, such as sample rate conversion quality, velocity response, sample storage, search-related parameters, and so on.

To open the Sampler Preferences window, do one of the following

- In the Parameter window, open the Options pop-up menu and choose Preferences.

- In the Instrument Editor, choose Edit > Preferences.



- *Sample Rate Conversion pop-up menu:* Determines the interpolation quality used by the EXS24 mkII. Choose Best to maintain the highest possible sound quality when transposing.
- *Sample Storage pop-up menu:* Determines the sample format handling method used by the EXS24 mkII.
  - *Original:* Loads samples into RAM at their original bit depth. These are converted to the internal 32-bit floating point format of the host application on playback.
  - *32 Bit Float:* Samples are stored and loaded in this format. This eliminates the need for any real time conversions, meaning that the EXS24 mkII handles samples more efficiently and can play back more voices simultaneously.

  Note: This requires twice as much RAM for 16-bit samples, and a third more RAM for 24-bit samples.
- *Velocity Curve slider:* Determines how the EXS24 mkII responds to incoming velocity values. Negative values increase the responsiveness to soft key strikes, and positive values decrease responsiveness.
- *"Search samples on" pop-up menu:* Determines the locations that the EXS24 mkII will search for instrument samples. You can choose from:
  - *Local Volumes:* Storage media (hard disks and optical mechanisms) attached to or installed in the computer.
  - *External Volumes:* Storage media accessible over a network.
  - *All Volumes:* Both internal and network media are scanned for appropriate data.

  Note: Choosing External Volumes or All Volumes may result in a dramatic increase in the time required by the EXS24 mkII to find and load sampler instruments and files.

- *"Read root key from" pop-up menu:* Sets the method used by the EXS24 mkII to determine the root key of loaded audio files. You can choose from the following:
  - *File/filename:* Initially reads information about the root key from the audio file itself (in the header of the AIFF or WAVE file) when loading it into a zone. If no information of this type exists in the file header, a smart analysis of the filename may detect a root key. If this second method doesn't provide any useful results, C3 will be used as the default root key in the zone.
  - *Filename/file:* As above, but the filename is read before the header.
  - *Filename only:* Reads from the filename only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.
  - *File only:* Reads from the file header only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.
- *"Root Key at File Name Position" pop-up menu:* Normally, the EXS24 mkII intelligently determines the root key from the file header of the loaded audio file. On occasion, you may want manual control over this parameter, if you feel that the root key is not being properly determined.
  - *Auto:* Provides a smart analysis of numbers and keys from the file name. A number in the file name can be recognized, regardless of its format—60 or 060 are both valid. Other valid numbers can range between 21 and 127. Numerical values outside of these are generally just version numbers. A key number is also a valid possibility for this use—C3, C 3, C_3, A-1, A-1 or #C3, C#3, for example. The possible range is C-2 up to G8.
  - *Numeric value:* There may be cases where a sound designer has used multiple numbers in a filename, which is common with loops, with one value being used to indicate tempo—"loop60-100.wav," for example. In this situation, it isn't clear which, if either, of the numbers indicates a root key or something else: 60 or 100 could indicate the file number in a collection, tempo, root key, and so on. You can set a value of 8 to read the root key at position (letter/character) eight of the filename—namely, the 100 (E6). Alternatively, setting a value of 5 will select the 60 (C3) as the root key position.
- *Previous Instrument and Next Instrument:* Determine which MIDI event type and data value will be used for selection of the previous or next instrument.

Important: These commands are unique to the EXS24 mkII and are separate from the global Previous/Next Plug-In Setting or EXS Instrument commands. Therefore, you should make sure that you do not assign the same MIDI event for both. If you do this, both commands will execute, which may result in unexpected behavior.

  - Choose the desired MIDI event type in the Previous Instrument and Next Instrument pop-up menus. Choices include: Note, Poly Pressure, Control Change, Program Change, Channel Pressure, and Pitch Bend. In the field next to each pop-up menu, you can enter either the note number or the value of the first data byte. When Control Change is chosen, the number field determines the controller number.

- *"Giga convert includes release trigger" checkbox:* Determines whether or not the release trigger function of the Gigasampler format will be performed by the EXS24 mkII.
- *"Ignore release velocity" checkbox:* Also refers to the release trigger function of the Gigasampler format, and it should always be selected for this purpose. Regardless of whether or not your keyboard is able to send release velocity, you would want your samples played by the release trigger function to be louder or softer than the original sample, or at the same volume, regardless of the initial velocity. When playing with release trigger, you would want the release velocity value to have the same value as the initial velocity value. To accomplish this, you can switch off release velocity.
- *"Keep common samples in memory when switching projects" checkbox:* Determines whether or not the samples commonly used (shared) by two open project files are reloaded when you switch between projects.

Configuring EXS24 mkII Virtual Memory

Multi gigabyte sample libraries are commonplace these days, delivering incredibly detailed and accurate instrument sounds. In many cases, these sample libraries are too large to fit into your computer's random-access memory (RAM).

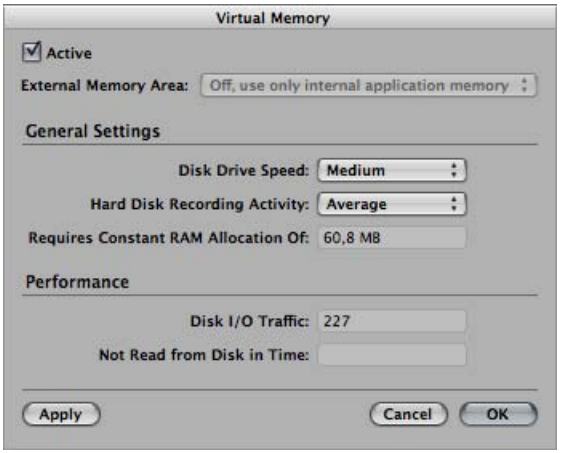
In order to use these huge sampler instruments, the EXS24 mkII can use a portion of your hard drive as virtual memory. When you turn on the EXS24 mkII's virtual memory, only the initial attacks of audio samples are loaded into the computer's RAM; the rest of the sample is streamed in real time from the hard drive.

For systems with 5 GB or more of RAM, and running the 32-bit version of the application, the virtual memory functions allow the EXS24 mkII to address its own memory. Systems running the 64-bit version of the application automatically address all available memory. See Advanced EXS24 mkII RAM Management.

You can configure the EXS24 mkII's virtual memory feature in the Virtual Memory window.

To open the Virtual Memory window

- Open the Options pop-up menu in the Parameter window and choose Virtual Memory.



- *Active checkbox:* Click to activate the EXS24 mkII's virtual memory feature.
- *External Memory Area pop-up menu:* 32-bit version only: Specifies the way memory (external to the RAM used by the application) is allocated. See Advanced EXS24 mkII RAM Management.
- *Disk Drive Speed pop-up menu:* Specifies the speed of your hard drive; if you have a 7200-rpm or faster hard drive for your audio samples, select Fast. If you are using a 5400-rpm laptop drive for your audio samples, select Medium. Generally, you will not need to use the Slow setting with any modern Macintosh.
- *Hard Disk Recording Activity pop-up menu:* Specifies overall hard disk usage—how much recording and streaming of non sampler-related audio you are doing. For example, if you are recording entire drum kits using over a dozen microphones, streaming live guitars and bass, recording choirs, and so on, you would set your hard disk recording activity to Extensive. On the other hand, if your projects consist mostly of software instrument playback, with perhaps a recorded instrument or vocal or two, you would set your hard disk recording activity to Less. If you are unsure, choose Average.
- *"Requires Constant RAM Allocation of" field:* Shows the memory requirements of the above parameters. The slower your hard drive and the higher your hard disk recording activity, the more RAM you will need to allocate to virtual memory.
- *Performance section:* Shows the current disk I/O traffic and the data not read from disk in time. If these numbers start rising, the EXS24 mkII may glitch when trying to stream your samples from disk. If you notice these values rising to high levels, you should change the general settings to free up additional RAM for virtual memory use. If you continue to see high Performance display values, and hear audio glitches, you should consider installing more RAM in your Macintosh.

Advanced EXS24 mkII RAM Management

32-bit applications can only address approximately 4 GB of RAM. The EXS24 mkII can address its own memory space in the 32-bit version of Logic Express. This enables the EXS24 mkII to use any available RAM in the system— outside the memory used by the application and operating system. You must have at least 5 GB of RAM installed in your computer to use this feature.

The 64-bit version of Logic Express has virtually no RAM restriction, and automatically addresses all available memory in the system. This enables the use of huge sampler instruments.

To set advanced memory options for the EXS24 mkII in 32-bit mode

1. Choose Options > Virtual Memory in the EXS24 mkII Parameter window.
2. In the Virtual Memory window, select the checkbox labeled Active.
3. In the External Memory Area pop-up menu, choose the required option.

Note: The External Memory Area pop-up menu in the Virtual Memory window is dimmed in the 64-bit version of Logic Express as the application directly addresses all available RAM. If you have enough physical RAM to hold all the samples for a project, then you will see somewhat better performance by disabling the Active checkbox. In projects with lots of audio tracks playing, and relatively few EXS24 instruments, this may have a noticable effect on performance. If the Active checkbox is disabled, and there is insufficient RAM to hold all samples, then the application will swap data to and from the disk, which will seriously affect performance. Turning off the Active checkbox will also increase project load times, so in most cases, it's best to leave it enabled.

The amount of RAM available for use by the EXS24 mkII is determined by several factors, including:

- The amount of physical RAM installed in your computer.
- How much RAM other open applications and the operating system are using.
- How much RAM Logic Express is using. This can be affected by the number and size of audio files in the project, and other plug-ins used. Sampler plug-ins not made by Apple, in particular, may significantly affect the amount of RAM that Logic Express uses.

Using the VSL Performance Tool in EXS24 mkII

The EXS24 mkII features an additional interface for the Vienna Symphonic Library Performance Tool. The Performance Tool software provided by VSL needs to be installed to allow access to this interface. For details, see the VSL documentation.

Klopfgeist

10

Klopfgeist is an instrument that is optimized to provide a metronome click in Logic Express.

Klopfgeist is inserted in the Logic Express instrument channel strip 256 by default and is used to generate the MIDI metronome click.

Klopfgeist can also be inserted in any other instrument channel strip in Logic Express for use as an instrument.

This chapter covers the following:

- Using the Klopfgeist Parameters (p. 242)

Using the Klopfgeist Parameters

An examination of Klopfgeist's parameters shows that it is a synthesizer designed to create metronome click sounds.

- *Trigger Mode buttons:* Click Mono to operate Klopfgeist as a monophonic instrument, or click Poly to operate it as a polyphonic (4 voice) instrument.
- *Tune knob and field:* Tunes Klopfgeist in semitone steps.
- *Detune knob and field:* Fine-tunes Klopfgeist in cents (1/100th of a semitone).
- *Tonality slider and field:* Changes the sound of Klopfgeist from a short click to a pitched percussion sound—similar to a wood block or claves.
- *Damp slider and field:* Controls the release time. The shortest release time is reached when *Damp* is at its maximum (1.00) value.
- *Level via Vel slider and fields:* Determine the velocity sensitivity of Klopfgeist. The upper half of the two-part slider determines the volume for maximum velocity, and the lower half determines the volume for minimum velocity. Drag the area between the two slider segments to move both simultaneously.

Ultrabeat

11

Ultrabeat is a synthesizer that is designed to create percussive sounds and polyphonic rhythms. It also incorporates a powerful, integrated step sequencer, which you can use to create polyphonic rhythmic sequences and patterns.

This chapter covers the following:

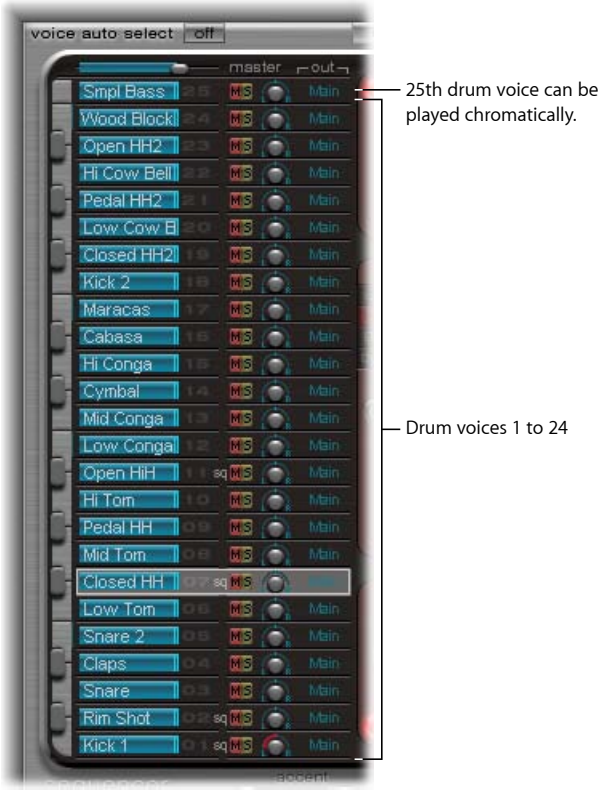
- Getting to Know the Structure of Ultrabeat (p. 244)
- Loading and Saving Ultrabeat Settings (p. 246)
- Getting to Know the Ultrabeat Interface (p. 247)
- Getting to Know Ultrabeat's Assignment Section (p. 247)
- Importing Sounds and EXS Instruments into Ultrabeat (p. 251)
- Getting to Know the Ultrabeat Synthesizer Section (p. 254)
- Getting to Know the Ultrabeat Oscillators (p. 256)
- Getting to Know the Ultrabeat Oscillator Parameters (p. 257)
- Using Oscillator 1 in Ultrabeat (p. 258)
- Using Oscillator 2 in Ultrabeat (p. 262)
- Using the Ultrabeat Ring Modulator (p. 267)
- Using the Ultrabeat Noise Generator (p. 268)
- Getting to Know the Ultrabeat Filter and Distortion Section (p. 269)
- Using Ultrabeat's Multimode Filter (p. 270)
- Using Ultrabeat's Distortion Circuit (p. 272)
- Using the Ultrabeat Output Section (p. 273)
- Working with Modulation in Ultrabeat (p. 279)
- Working with the Ultrabeat Step Sequencer (p. 289)
- Automating Parameter Values in Ultrabeat's Step Sequencer (p. 299)
- Exporting Ultrabeat Patterns as MIDI Regions (p. 303)
- Using MIDI to Control Ultrabeat's Sequencer (p. 304)

- Ultrabeat Tutorial: Introduction (p. 305)
- Ultrabeat Tutorial: Creating Kick Drums (p. 305)
- Ultrabeat Tutorial: Creating Snare Drums (p. 310)
- Ultrabeat Tutorial: Creating Toms and Tonal Percussion (p. 316)
- Ultrabeat Tutorial: Creating Hi-Hats and Cymbals (p. 316)
- Ultrabeat Tutorial: Creating Metallic Sounds (p. 317)
- Ultrabeat Tutorial: Creating Extreme Sounds (p. 318)
- Ultrabeat Tutorial: Programming in Building Blocks (p. 318)

Getting to Know the Structure of Ultrabeat

Most software synthesizers offer one synthesizer per plug-in instance. Ultrabeat, however, places 25 independent synthesizers at your disposal. These synthesizers—known as *drum voices* in Ultrabeat—are optimized for the generation of drum and percussion sounds.

The distribution of drum voices across the MIDI keyboard is simple—a single drum voice is assigned to the first 24 MIDI keys, starting from the bottom. The 25th drum voice is assigned to the keys above—from the 25th key—allowing this sound to be played chromatically.



You can compare Ultrabeat with a drum machine that features 24 drum pads, plus a built-in multioctave keyboard, which can be used for polyphonic accompaniments, or bass or melody lines.

Ultrabeat's 24 drum pads are assigned to the bottom 24 keys of a standard MIDI keyboard (spanning MIDI notes C1 to B2). This is compatible with the widely adopted GM (General MIDI) MIDI Drum note mapping standard. The lowest note in the range for the 25th synthesizer begins at C3. All notes above—and C3 itself—can be played chromatically.

Note: If your MIDI keyboard is limited to two octaves or does not support transposition, you can use the Transpose parameter of your host application to shift incoming MIDI notes up or down one or more octaves.

Important: For clarity, and to maintain the drum machine analogy, this guide refers to the independent synthesizers as drum sounds that, combined, form a drum kit.

Loading and Saving Ultrabeat Settings

Ultrabeat settings are saved and loaded in the same way as other Logic Express instruments. For more information, see the Logic Express documentation.

An Ultrabeat setting contains:

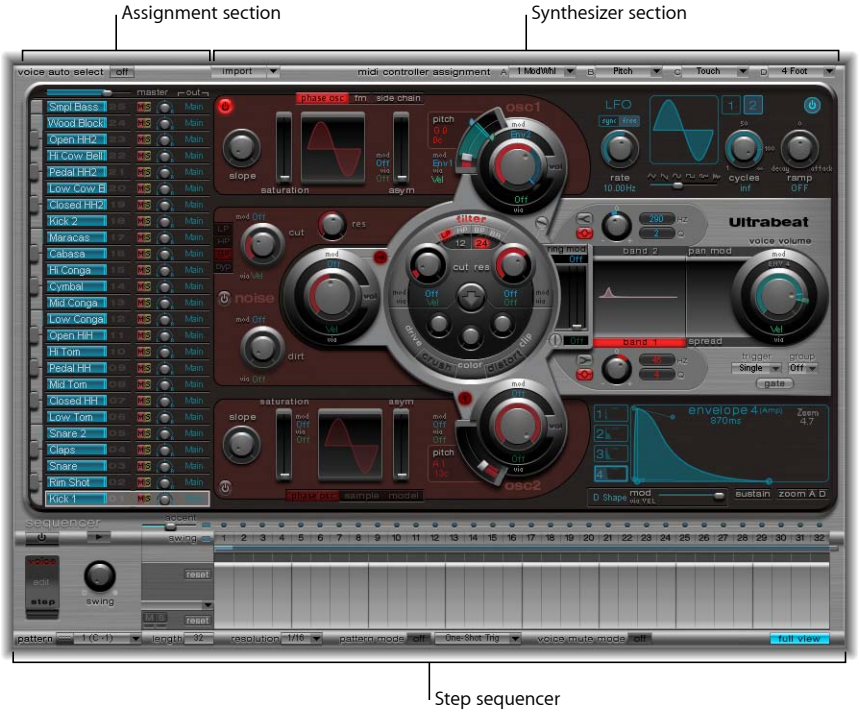
- The drum kit, which consists of 25 drum sounds, including assignment and mixer settings
- The complete parameter settings of all 25 sounds
- The sequencer settings and all patterns, including step automation, trigger, velocity, and gate information for all 25 sounds

The joint recall of all this information when loading an Ultrabeat setting is extremely practical, because the musical effect of the patterns—especially those with sequenced gate and velocity parameters—is often tightly tied to the tone and impact of the sounds being used.

Note: When you save a drum kit with the Settings menu, only the location of the sample is saved with the setting. An Ultrabeat setting doesn't actually save the audio files themselves—merely a reference to their hard disk location. If you load a setting that contains a reference to a sample that has been moved or erased, a dialog prompts you to specify or find the sample. To avoid this problem, it is highly recommended that you use the Finder to create and manage a dedicated Ultrabeat sample folder—for all sounds and kits.

Getting to Know the Ultrabeat Interface

Ultrabeat's user interface is divided into three main sections.



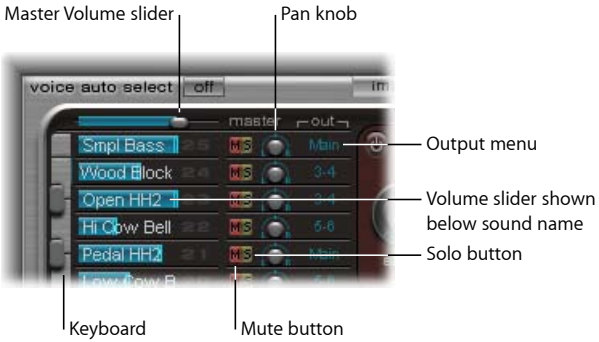
- *Assignment section:* Displays all drum sounds in a drum kit, allowing you to select, rename, and organize them. It also includes a small mixer, used to adjust the level and pan position of each sound. See Getting to Know Ultrabeat's Assignment Section.
- *Synthesizer section:* Used to create and shape individual drum sounds. The parameters of the drum sound selected in the Assignment section are displayed in the Synthesizer section. See Getting to Know the Ultrabeat Synthesizer Section.
- *Step sequencer:* Used to create and control sequences and patterns. A sequence triggers a single drum sound, and can consist of up to 32 steps. A pattern contains the sequences for all 25 sounds. You can trigger and control sounds with the step sequencer—in place of, or in addition to—MIDI notes entering Ultrabeat (from your host application). See Working with the Ultrabeat Step Sequencer.

Getting to Know Ultrabeat's Assignment Section

The Assignment section displays all sounds in a drum kit. It allows you to:

- Select, organize, and name sounds.
- Import sounds from other Ultrabeat settings or EXS instruments.

- Set relative levels and pan positions for each sound.
- Mute or solo sounds in the drum kit.



- *Master (Volume) slider:* Controls the levels of all drum sounds in the kit—the overall mix level of all drum sounds.
- *Volume slider:* Sets the volume of each sound. All sound levels are indicated by blue sliders that provide a complete overview of all relative levels within the kit. Drag the blue slider to set the volume for the desired sound.
- *Keyboard:* Acts as a display when appropriate MIDI information is received. You can also click the keys to play the sound on the corresponding row.
- *M(ute) button:* Mutes individual sounds in a drum kit. You can simultaneously mute several sounds or sequences by clicking the appropriate Mute buttons.
- *S(olo) button:* Allows you to listen to sounds in isolation. You can simultaneously solo several sounds or sequences by clicking the appropriate Solo buttons.
- *Pan knob:* Controls the placement of sounds in the stereo field (panorama).
- *Output menu:* Used to independently route each drum sound to individual outputs, or output pairs. Ultrabeat features eight separate stereo and mono outputs when inserted as a multi-output instrument.
  - Drum sounds that are routed to an output pair other than Main (1–2) are automatically removed from the main output channel strips.
  - Choosing an output pair other than Main (1–2) routes the sound to an aux channel strip.

Playing and Selecting Ultrabeat Drum Sounds

The 25 sounds of an Ultrabeat drum kit are mapped to the onscreen keyboard at the left side of the Ultrabeat interface. Sounds start from the bottom of the onscreen keyboard and correspond to note values on a connected MIDI keyboard, starting at C1.

To play a sound

Do one of the following:

- Play a note on a connected MIDI keyboard. As mentioned, the bottom sound is mapped to MIDI note C1.
- Click a key on the onscreen keyboard to trigger the sound on the adjacent row.

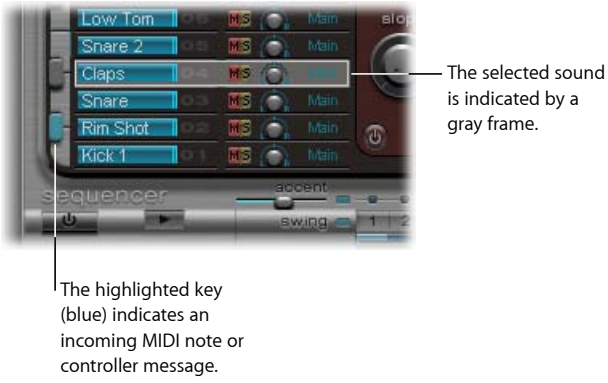
The corresponding key on the onscreen keyboard to the left of the sound name turns blue when it is clicked or when it receives appropriate MIDI information.

To select a sound

- Click the name of the sound in the Assignment section.

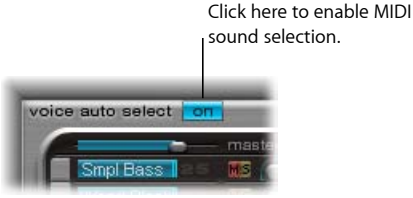
You can recognize the selected sound by the gray frame that appears around the entire assignment row. The parameters of the selected sound are shown in the Synthesizer section to the right. See Getting to Know the Ultrabeat Synthesizer Section.

The example below shows drum sound 2 being played (as indicated by the blue key) while drum sound 4 is selected (the gray frame).



To select a sound by playing your MIDI keyboard

- Activate the Voice Auto Select button in the upper-left corner of the Ultrabeat window.



Naming, Swapping, and Copying Ultrabeat Drum Sounds

This section outlines the steps required to rename and rearrange the positions of individual sounds within a drum kit.

To rename a sound

1. Double-click the name of a sound to open a text entry field.
2. Type in the name and press Return, or click anywhere outside the text entry field, to complete the naming operation.

Swapping and copying drum sounds within an Ultrabeat kit can be achieved with a drag-and-drop or shortcut menu operation.

To swap or copy drum sounds using drag and drop

1. Select the sound name (not on a button or menu).

2. Drag it to the desired position. The target is shaded as you drag across the list of sound names.
   - A simple drag and drop swaps the two drum sounds (including Mixer settings: volume, pan, mute, solo, and output configuration). Sequences are*not* swapped.
   - Command-drag to swap the two drum sounds and sequences.
   - Option-drag to copy the sound. Sequences are*not* copied.
   - Command-Option-drag to copy the sound and sequences.

To swap or copy drum sounds with a shortcut menu command

1. Control-click or right-click the sound name.

2 Choose one of the following commands from the shortcut menu:

- *Copy (Voice & Seq):* Copies the selected sound, including mixer settings and all sequences, to the Clipboard.
- *Paste Voice:* Replaces the selected sound with the sound from the Clipboard but does not replace existing sequences.
- *Paste Sequence > (submenu):* Enables you to replace all, or individual sequences, of the target drum sound. Sound parameters are not affected.
  - Pasting a single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This allows you to copy sequences into any of the 24 possible pattern locations.
  - Pasting all sequences replaces all sequences.
  - In situations where a sound only has several sequences (not all 24 are used), choosing "Paste Sequence > all" will place these sequences into the same positions; sequence 5 (in the Pattern menu) will be pasted to the same position in the new sound, for example. If a sequence already exists at this location in the target sound, it will be replaced. If no sequence exists at this location, the copied sequence will be added to the drum sound.
- *Swap with Clipboard:* Exchanges and replaces the selected sound with the sound from the Clipboard.
- *Init > (submenu):* Opens a submenu that contains a few starting-point (Init) sounds. Select one of these to replace the target drum sound.
  - The Sample Init sound initializes the filter and pitch parameters to neutral settings, which are ideal as a starting point for programming sample-based drum sounds.

These commands affect only the selected drum sound. The sequence and sound data of the other 24 sounds are unaffected.

Note: The shortcut menu commands Paste and "Swap with Clipboard" require an initial Copy command—to place data in the Clipboard—before they can be used.

Importing Sounds and EXS Instruments into Ultrabeat

Ultrabeat can directly import EXS instruments, in addition to loading audio samples and its own settings. This facilitates enhanced sound design and rhythm processing options for EXS instruments. A further bonus is the ability to play and control EXS drum kits with Ultrabeat's intuitive drum mixer layout.

You use Ultrabeat's import list to add drum sounds and sequences from other Ultrabeat settings or EXS instruments to your currently active Ultrabeat drum kit (see Adding Sounds and Sequences to Your Ultrabeat Drum Kit). You can also import an entire EXS instrument by dragging an EXS instrument file from the Finder directly onto the Assignment section (see Importing Entire EXS Instruments into Ultrabeat).

Adding Sounds and Sequences to Your Ultrabeat Drum Kit

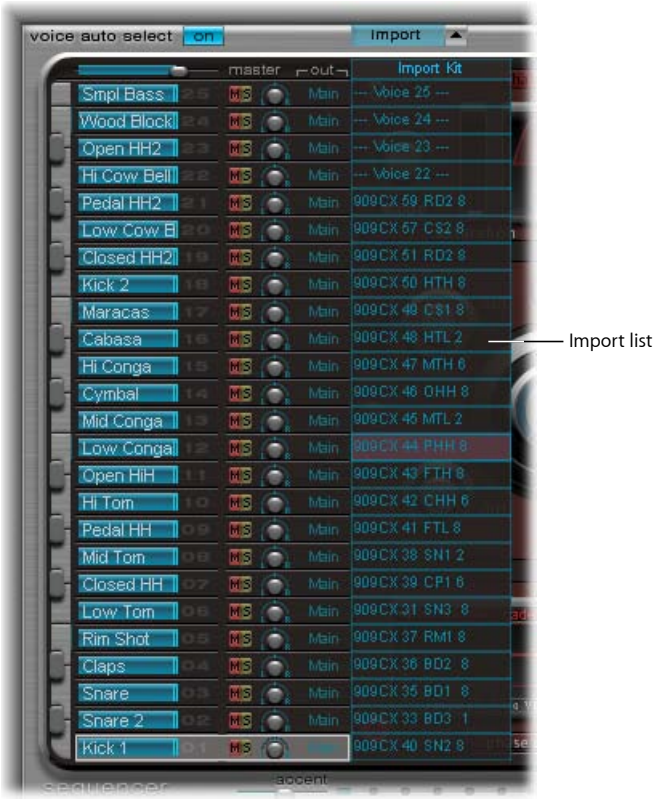
You can use Ultrabeat's import list to add drum sounds and sequences from other Ultrabeat settings or EXS instruments to your currently active Ultrabeat drum kit.

To open an Ultrabeat setting or EXS instrument with the import list

1 Click the Import button near the upper-left corner of the interface.

2 Choose the source Ultrabeat setting or EXS sampler instrument in the file selector, then click Open.

A list of all sounds found in the selected setting, or samples in the EXS instrument, is shown to the right of the Assignment section Mixer.



Note: If you import EXS instruments that include more than 25 sample zones, you can navigate through pages (of 25 zones) by using the up and down arrows to the left and right of the EXS instrument name at the top of the import list.

There are two methods for transferring sounds and sequences from the import list into your drum kit.

To drag sounds into your drum kit

- Drag the sound name from the import list into the desired row in the Mixer section.

  Note: Hold down the Command key to include all sequences.

To transfer sounds using shortcut menu commands

1. Control-click (or right-click) the sound name in the import list.
2. Choose Copy (Voice & Seq) in the shortcut menu.

   This copies the selected sound and its sequences to the Clipboard.
3. Control-click (or right-click) the sound you want to replace in the current drum kit, then choose one of the following shortcut menu commands:

   - *Paste Voice:* Replaces the selected sound with the sound from the Clipboard but does not replace existing sequences.
   - *Paste Sequence > (submenu):* Opens a submenu that enables you to replace all, or individual sequences, of the target drum sound. Sound parameters are not affected.
     - Pasting a single sequence replaces the currently active sequence (as set in the Pattern menu) of the target drum sound. This allows you to copy sequences into any of the 24 possible pattern locations.
   - *Swap with Clipboard:* Exchanges and replaces the selected sound (and associated sequence) with the sound from the Clipboard.

Importing Entire EXS Instruments into Ultrabeat

You can import an entire EXS instrument into Ultrabeat by dragging an EXS instrument file from the Finder directly onto the Assignment section.

Ultrabeat reproduces the EXS layout as closely as possible. Layered EXS zones are set up as layered drum sounds, using the sample playback mode of Oscillator 2 (see Using Ultrabeat's Sample Mode).

Note: This method does not allow paging through the EXS instrument if it contains more than 25 sounds (samples). Ultrabeat will map only sample zones and layers that fall within Ultrabeat's drum sound range of C1 to C3. All other samples (zones) are ignored.

Getting to Know the Ultrabeat Synthesizer Section

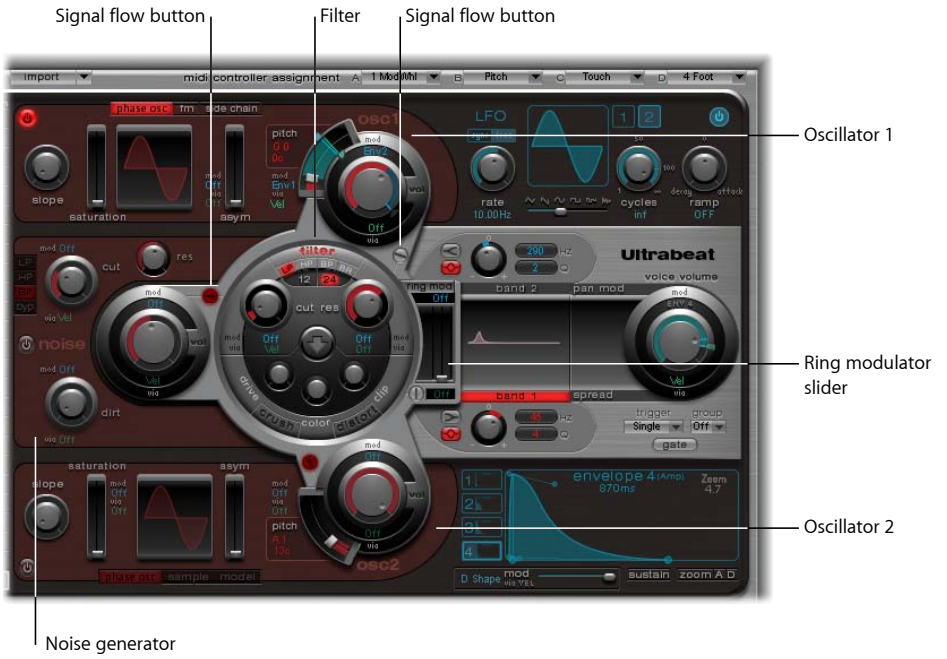
Ultrabeat's sound engine is optimized for creating electronic and acoustic drum and percussion sounds. It combines several synthesis approaches—phase distortion, sample playback, FM (frequency modulation), and physical modeling—to create tones. You can even use an audio side-chain input as a sound source, if desired. The sound engine provides truly comprehensive modulation functions—allowing nearly every aspect of Ultrabeat to be modulated.

The Synthesizer section is the heart and soul of Ultrabeat. Each drum sound in a kit is an independent synthesizer and has its own set of synthesizer parameters—its own Synthesizer section.

The interface and signal flow of Ultrabeat's synthesis engine are based on classic synthesizer designs. If you're totally new to synthesizers, it might be best to start with Synthesizer Basics, which will introduce you to the fundamentals and terminology of different synthesis systems.

If you look at the Synthesizer section from left to right, you'll recognize the layout and signal flow of a subtractive synthesizer. First, the basic tonal material is created by the oscillators, noise generator, and ring modulator. A filter then takes away certain frequencies from the raw sound, followed by volume shaping—envelopes.

Note: Whereas the structure and layout mirrors classic subtractive synthesizer designs, Ultrabeat incorporates a number of different tone generation (synthesis) methods, including frequency modulation, component modeling, sample playback, and phase distortion. These provide unique qualities that massively expand the range of sounds you can create.



The details of Ultrabeat's functions and their importance become more apparent when you look at the three-dimensional nature of the interface and recognize the different levels from front to back. The following descriptions refer to the third dimension, so keep this in mind while reading and exploring Ultrabeat's interface.

The large, round, elevated Filter (and Distortion) section is in the center. Its placement and design are both symbolic and practical, as the filter plays a central role in Ultrabeat. The Filter is discussed in Getting to Know the Ultrabeat Filter and Distortion Section.

The Filter receives its signal from the following sound sources: Oscillator 1, Oscillator 2, the noise generator, and the ring modulator. The outputs of these sources are represented by the three round objects and the smaller, rectangular ring modulator section to the right, that surround the Filter.

One level down—from front to back—each of the sound source output objects provides controls that determine how modulation sources, such as the LFO and envelopes, affect the sound source. All modulation sources are discussed in Working with Modulation in Ultrabeat.

Each of the sound sources that adjoins the filter also features a small signal flow button (red, when active). This is used to determine (and indicate) whether the signal of the associated sound source should proceed through the Filter or bypass it—before being routed to Ultrabeat's Output section.

The Output section is shown to the right. Signals sent from the Filter can pass through two equalizers and a stage for stereo expansion or panoramic modulation. You can also set up the initial output level and trigger behavior in this section. See Using the Ultrabeat Output Section.

The output of the drum sound is then sent to the Assignment section mixer (see Getting to Know Ultrabeat's Assignment Section).

Getting to Know the Ultrabeat Oscillators

The Ultrabeat oscillators are used to generate waveforms. Oscillator 2 can use a sample in place of a waveform. The signal of one, or both, oscillators is then sent to other portions of the synthesizer engine for shaping, processing, or manipulation.

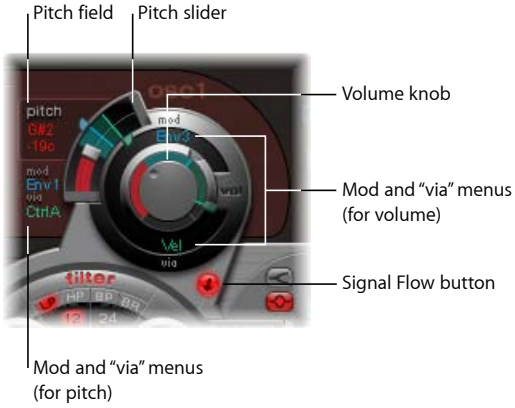
Other sound sources include a separate noise generator and ring modulator, which can produce additional signals to those generated by the oscillators. See Using the Ultrabeat Ring Modulator and Using the Ultrabeat Noise Generator.

Before the oscillator parameters are discussed, a few of the special features available to you in Ultrabeat's oscillator section should be pointed out.

- Oscillator 1 can be frequency-modulated by Oscillator 2, for FM synthesis sounds.
- Oscillator 2 can be ring-modulated with Oscillator 1.
- Oscillator 2 can use an audio file (a sample), in place of a synthetic waveform. The sample is output as the Oscillator 2 signal.

Getting to Know the Ultrabeat Oscillator Parameters

This section provides an overview of the parameters available to both oscillators. Further sections discuss parameters that are unique to each oscillator.



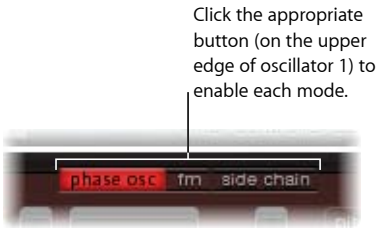
- *Oscillator on/off button:* Click the button (at the top left of Oscillator 1 or bottom left of Oscillator 2) to activate or deactivate each independently. A red button indicates an active oscillator.

  Note: When programming a drum sound, you can enable or disable individual sound sources with the appropriate on/off buttons, allowing you to hear components of the sound in isolation.

- *Volume knob:* Sets the level of Oscillator 1 or 2.
  - Volume can be modulated by the sources found in the "mod" and "via" menus.
- *Pitch slider and field:* Sets the oscillator pitch—in semitone steps. The pitch value is displayed in the field to the left of the slider. Drag vertically on the field to adjust.
  - Press Shift to adjust the pitch in cent intervals (1 cent = 1/100th semitone).
  - Pitch can also be modulated by the sources found in the "mod" and "via" menus.
- *Signal Flow button:* Determines whether the signal of the associated oscillator will be routed through the Filter, or sent directly to the EQ (in the Output) section. When active, the button turns red and an arrow indicates the direction of the signal flow.
- *"Mod" and "via" menus:* Determine the modulation sources for oscillator pitch and level. See Working with Modulation in Ultrabeat.

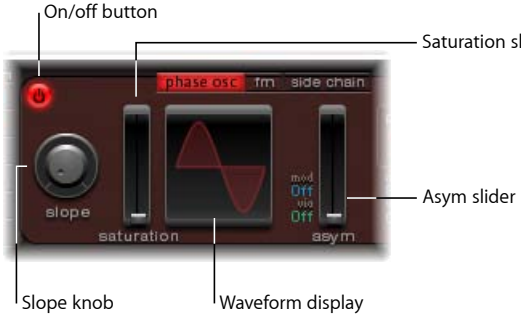
Using Oscillator 1 in Ultrabeat

Oscillator 1 can be switched between three different modes. In essence, this enables different types of synthesis engines: phase oscillator, fm, and side chain (external audio input), which extends your sonic palette significantly. Each mode offers different parameters and features.



Using Oscillator 1's Phase Oscillator Mode in Ultrabeat

The waveform of the phase oscillator can be reshaped into almost any basic synthesizer waveform. The effect of parameter changes are immediately reflected in the waveform display within the Oscillator section.



- *Slope knob:* Determines the slope or steepness of the waveform. The higher the value, the steeper the waveform. The resulting sound takes on an increasingly nasal character as the incline becomes more vertical.
- *Saturation slider:* Increases the gain, eventually causing the waveform to clip. Higher values result in a distortion of the waveform shape, making it more rectangular. This results in a corresponding increase in odd-numbered overtones.
- *Asym(metry) slider:* Changes the waveform angle. Higher values skew the waveform toward a sawtooth wave, making the sound more edgy. Asym can be modulated by the sources found in the "mod" and "via" menus. This allows you to create dynamic sound changes at the oscillator level. For more information, see Working with Modulation in Ultrabeat.

Creating Classic Waveforms in Ultrabeat's Phase Oscillator Mode

The basic waveforms of classic analog synthesizers can be easily reproduced with the phase oscillator: sine, rectangular, and sawtooth waves will result from different Slope, Saturation, and Asym parameter value combinations.

For example, setting Slope and Saturation to their maximum values, and Asym to the minimum value will result in a classic square wave. Setting Slope to –0.20, Saturation to the minimum, and Asym to the maximum value, will result in a sawtooth wave. Setting all three parameters to 0 values will cause the oscillator to produce a sine wave. See the table below for an overview of the tonal qualities of each basic waveform.

Waveform	Basic tone	Comments
Rectangular	Nasal sounding	Great for reed instruments, synth blips, basses
Square	Hollow and woody sounding	Useful for basses, clarinets, and oboes. The pulse width of (oscillator 2 and 3) square waveforms can be smoothly scaled between 50% and the thinnest of pulses.
Sawtooth	Warm and even	Useful for strings, pads, bass, and brass sounds
Triangle	Sweet sounding, softer than sawtooth	Useful for flutes, pads
Sine	A pure tone	The sine wave of Oscillator 1 can be frequency-modulated by Oscillator 2. This kind of frequency modulation forms the basis of FM synthesis.

Using Ultrabeat's FM (Frequency Modulation) Mode

Whereas the phase oscillator is well suited to simulations of analog waveforms and analog-style sounds, FM mode is well suited to the creation of bell-like digital tones and metallic sounds.

The principle of frequency modulation (FM) synthesis was developed in the late 1960s and early 1970s by John Chowning. It was popularized by Yamaha's range of DX synthesizers in the 1980s. Ultrabeat can't be compared with the DX series in the discipline of pure FM synthesis, but it can certainly achieve some of the signature sounds of these instruments.

How Frequency Modulation Works

Put very simply, the frequency of one signal generator (oscillator) is altered, or modulated, by another signal generator. Positive, or higher, frequency values from the second generator, known as the*modulator*, increase the frequency of the first generator, known as the*carrier*. Negative, or lower, frequency values from the second generator decrease the frequency of the first.

In a synthesizer, this type of modulation takes place in the audio spectrum. Depending on the design of the instrument, you can hear the signals of either the carrier alone, being modulated by the modulator, or both oscillators. The interaction between the two oscillators alters the waveform signal of the carrier and introduces a number of new harmonics. This harmonic spectra can then be used as the source signal for further sound processing, such as filtering, envelope control, and so on. For further information, see Frequency Modulation (FM) Synthesis and EFM1.

Frequency Modulation in Ultrabeat

In FM mode, Oscillator 1 (the carrier) generates a sine wave. Its frequency is modulated by the waveform of Oscillator 2 (the modulator).

- When Oscillator 2 outputs a positive (or higher) frequency signal, the frequency of Oscillator 1 increases.
- When Oscillator 2 outputs a negative (or lower) frequency signal, the frequency of Oscillator 1 decreases.

The net effect of speeding up or slowing down the frequency of Oscillator 1 in each waveform cycle is a distortion of the basic wave shape. This waveform distortion also has the side benefit of introducing a number of new, audible, harmonics.

The more complex the Oscillator 2 waveform, the more partials will be created by increasing the FM Amount during the FM process. Keep an eye on the display to see how the sine wave takes on an increasingly complex shape.

Note: Oscillator 2 must be enabled if you want to use frequency modulation.

Important: The impact of any frequency modulations you may perform depends on*both* the frequency ratio and the modulation intensity of the two oscillators.

To set the frequency ratio and adjust the modulation intensity

1. Adjust the Pitch parameter values of one, or both, oscillators.
2. Adjust the amount (intensity) of frequency modulation with the FM Amount knob.

This parameter can be modulated by the sources found in the “mod” and “via” menus. For more information, see Working with Modulation in Ultrabeat.

Using Ultrabeat's Side Chain Mode

In Side Chain mode, Ultrabeat uses an external side-chain input as the source for Oscillator 1. This means that you can send the signal of any audio channel strip, any bus, or live input through Ultrabeat's filters, envelopes, LFO, and step sequencer. Using busses as sidechain sources makes it possible to route signals to the sidechain input from any channel strip type that offers busses as outputs or sends. This includes software instrument channel strips, aux channel strips, or a mix of multiple channel strips that are routed into a common aux (sub group), that has a bus as the output destination.

Here is an example of how you might take advantage of this feature: You could use an audio input from Oscillator 1, along with the synthesis engine of Oscillator 2, to create a part live audio, part synthesized drum sound.

Note: The side chain affects only the selected drum sound—Ultrabeat's other drum sounds and sequences behave as usual.

Here is another example of where this could be useful: You could use one drum sound in a kit to filter an external audio signal with a programmed groove.

Note: A side chain audio signal alone is not enough to trigger Ultrabeat. To hear the side-chained audio signal, Ultrabeat needs to be triggered by MIDI or the internal step sequencer.

To use Side Chain mode

1 Activate the "side chain" button for Oscillator 1.

2 Choose the channel strip that you want to use as the side chain input source from the Side Chain pop-up menu at the top of the plug-in window.

Using Oscillator 2 in Ultrabeat

Oscillator 2 can also be switched between three different types of synthesis engines: phase oscillator, sample, and model. Each mode offers different parameters and features.

Click the appropriate button on the lower edge of the Oscillator 2 section to enable each mode.

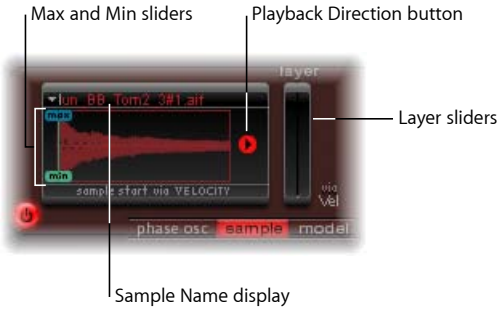
Using Oscillator 2's Phase Oscillator Mode in Ultrabeat

Oscillator 2's phase oscillator operates in a nearly identical fashion to the phase oscillator of Oscillator 1. The key difference is that Saturation can be modulated in Oscillator 2, rather than Asymmetry (in Oscillator 1). This means that when both oscillators are in phase oscillator mode, they can produce different sounds.

For details on parameters, see Using Oscillator 1's Phase Oscillator Mode in Ultrabeat.

Using Ultrabeat's Sample Mode

In Sample mode, Oscillator 2 uses an audio file as a sound source.



- *Sample Name display:* Used to load and unload samples or to display the loaded sample in the Finder. Click the arrow in the upper-left corner of the waveform display to open. See Loading Samples into Ultrabeat Oscillator 2.
- *Max/Min sliders:* Determine the start point of the sample—depending on the dynamics (incoming velocity level) of the performance. To adjust, drag the max/min labels to the left or right.
  - *Min:* Determines the start point of the sample at the minimum velocity level (velocity = 1).
  - *Max:* Determines the start point of the sample at the maximum velocity level (velocity = 127).

Note: If Min and Max are set to the same value, velocity has no effect on the sample start point.

- *Playback Direction button:* Changes the playback direction of the sample (forward or backward).
- *Layer sliders:* Both factory Ultrabeat samples and sounds imported from EXS instruments often consist of different layers that are dynamically switched by incoming MIDI note velocities. The precise sample layer that incoming velocity values switch to is determined by the green Layer slider on the right (min), or the blue slider on the left (max).
  - The green Min slider on the right determines which layer is triggered at a MIDI note velocity = 1.
  - The blue Max slider on the left determines which layer is triggered at a MIDI note velocity = 127.

  Note: If you have loaded a single sample that does not have multiple layers, the Vel Layer slider has no effect.

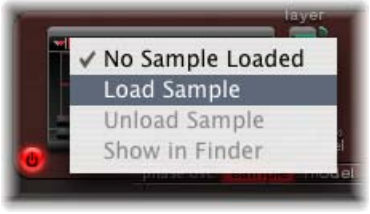
Loading Samples into Ultrabeat Oscillator 2

A selection of multilayer drum and percussion samples is included on your installation discs. These were especially created for Ultrabeat and its function set. You can also load your own samples in AIFF, WAV, CAF, or SDII stereo interleaved format.

Note: The velocity layering function is not available for user-loaded samples.

To load a sample into Oscillator 2

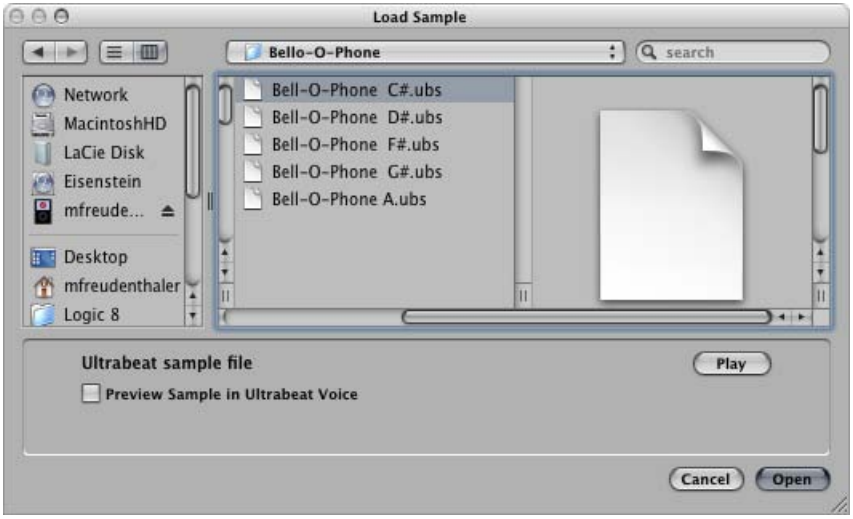
1. Click the arrow in the upper-left corner of the waveform display, then choose Load Sample from the pop-up menu.



2. In the Load Sample window, browse to the desired audio file, and do one of the following:
   - Click Open to load the chosen file into Oscillator 2.
   - Click Cancel to retain the current state.

Using Ultrabeat's Load Sample Preview Function

You can audition samples before importing them into Ultrabeat.



- *Play button:* Allows you to preview audio files (AIFF, WAV, SD2, CAF, UBS) before loading.
  - Click the Play button to loop playback of the currently selected sample file. The sample is played directly from the hard disk, with no manipulation: filters, EQ, envelopes, and other synthesizer parameters are ignored.
  - Click the button again to stop playback.
  - You can audition multiple files by clicking Play once, and then stepping through the files by pressing the Up Arrow and Down Arrow keys or by clicking each file.

  Note: Independent auditioning of all layers is not possible. In multilayer UBS files, the audition function plays the sample at a fixed velocity of 75%. Only the layer addressed by this velocity value will be played.
- *"Preview Sample in Ultrabeat Voice" checkbox:* Temporarily replaces the sample files in the currently selected drum sound, in the Assignment section. The drum sound can be triggered as usual (played notes, MIDI region events, or Ultrabeat sequencer events) while the Load Sample window is open and different files are being selected. The selected sample can be heard as part of the current drum sound, inclusive of all synthesizer processing.

Important: Any effects inserted into the Ultrabeat instrument channel strip are heard when you preview samples.

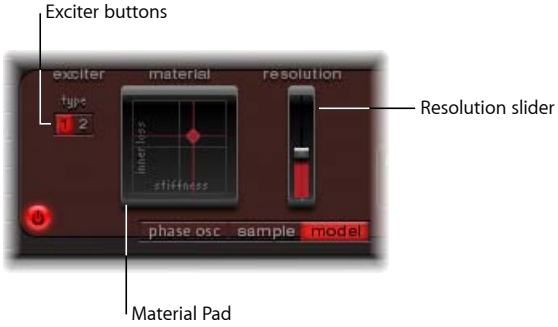
Using Ultrabeat's Model Mode

This mode employs a method of synthesis known as component modeling. This tone generation technique emulates the physical properties of an object, such as a guitar string. Further objects are used to stimulate the string—in essence, emulating the way that it is played: plucked, bowed, and so on. Although the term *string* is used, Model mode enables you to create sounds that have little in common with traditional stringed instruments.

The string is the element that is responsible for the basic tone. Ultrabeat offers parameters that allow you to adjust its material—what it's made of, in other words.

One of the two available exciters make the string vibrate (move) in different ways. The string itself doesn't make a sound unless it is stimulated, or excited.

The signal of the vibrating string is sent to the filter, amplifier, and so on in the Ultrabeat Synthesizer section.



- *Exciter buttons:* Choose one of the two contrasting exciters. Each offers different sound characteristics (Type 1 and Type 2).

  Note: In this context, an exciter is the agent or triggering device used to initiate the vibration of the string. Don't confuse it with the effect plug-in of the same name.

- *Material Pad:* Determines the basic tone of the string with the string Stiffness and damping (Inner Loss) parameters. See Using the Ultrabeat Material Pad.

- *Resolution slider:* Determines the precision of the calculation.
  - Higher values produce additional harmonics.
  - Lower values produce fewer harmonics or inharmonic spectra.

  Note: The higher the resolution (the more precise the calculation), the more CPU resources are required.

Using the Ultrabeat Material Pad

The combination of the Inner Loss and Stiffness parameter positions determine the string material and, therefore, the general timbre of your sound. Both parameters are simultaneously controlled by dragging the ball (which correlates to the x and y coordinates) within the*Material Pad*. The default pitch of the string is C3 (middle C).

In general synthesizer terms, this could be viewed as being similar to the waveform selector/generator in the oscillator section.

Note: Option-click the ball to reset all string parameters to their default values.

- *Inner Loss:* Emulates damping of the string, as caused by the string material—steel, glass, nylon, or wood. These are frequency-dependent losses that cause the sound to become more mellow during the decay phase.
- *Stiffness:* Sets the rigidity of the string. In reality, this is determined by the string material and diameter—or, to be more precise, by its geometrical moment of inertia.

Stiffer strings exhibit an inharmonic vibration, where overtones are not integer multiples of the base frequency. Rather, they have higher frequencies, which can make upper or lower notes sound somewhat out of tune with each other.

How Inner Loss and Stiffness Settings Can Change Ultrabeat's Tonal Color

Low Stiffness values, combined with low Inner Loss values, lead to metallic sounds.

Increasing the Stiffness makes the sound become more bell-like, or glass-like. Extreme Stiffness values turn the string into a solid metal rod.

Increasing the Inner Loss value, while maintaining a low Stiffness level, corresponds to nylon or catgut strings.

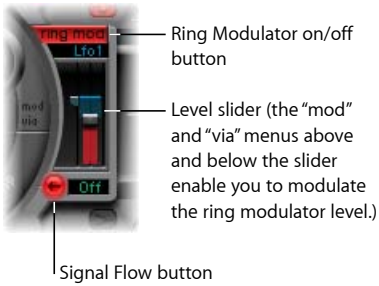
High Stiffness values, combined with high Inner Loss values, simulate wood-like materials.

Using the Ultrabeat Ring Modulator

Ring modulation is a powerful tool for the creation of inharmonic, metallic, bell-like sounds. Ultrabeat's ring modulator functions as an independent sound source—its signal can bypass or be sent into the filter, independent of the Oscillator 1 and 2 signals. Its volume can also be regulated.

Important: Although the ring modulator signal is independent of the signals generated by Oscillators 1 and 2, both oscillators need to be enabled if you want to use the ring modulator signal. As the ring modulator is reliant on the signals of both oscillators to produce an output signal, it is automatically muted when one of the oscillators is switched off.

The actual sound produced by the ring modulator is largely dependent on the parameter settings of both oscillators. In particular, the tuning relationships of each oscillator have a direct effect on the sound of the ring modulator signal. The individual levels of the oscillators, however, have no effect on the process (or output) of ring modulation.


ring mod
Lfo 1
mod
via
Off
Ring Modulator on/off button
Level slider (the "mod" and "via" menus above and below the slider enable you to modulate the ring modulator level.)
Signal Flow button

- *Ring Modulator On/Off button:* Enables or disables the ring modulator. When switched on, the label is highlighted.

  Note: If you want to hear the ring modulator's signal in isolation (in order to better judge your settings), temporarily set the volume of both oscillators to a value of 0.

- *Level slider:* Adjusts the output volume of the ring modulator.
- *"mod" and "via" menus:* Found above and below the Level slider, these determine the modulation source (and via source) for the Level parameter. When either source is active, small sliders (handles) appear on either side of the Level slider. See Working with Modulation in Ultrabeat.
- *Signal Flow button:* Determines the routing of the ring modulator signal. It is sent either to the filter (the signal flow button turns red), or directly to the EQ section (the signal flow button remains gray). The direction of the arrow on the signal flow button illustrates the routing.

  Note: The Signal Flow button determines how the ring modulator output signal is routed. It doesn't turn the ring modulator on or off.

Using the Ultrabeat Noise Generator

The Ultrabeat noise generator offers extensive features, enabling the creation of a wide range of percussive sounds and sound elements. The noise generator also has its own filter—which functions independently of the main Ultrabeat Filter, although this can also be used on the overall sound.

Technically, a noise signal contains all tonal frequencies, at a more or less equal volume level. As all frequencies in the spectrum are audible, it makes it difficult for human beings to isolate, and therefore discern, any tonality (pitch) in a noise signal. Despite this, or as a direct result of it, noise is an indispensable ingredient when creating drum sounds.

- *On/Off button:* Activates or deactivates the noise generator.

  Note: When programming a drum sound, you can turn the individual sound sources on or off with the On/Off buttons. You can also listen to the individual components of the sound separately this way, and remove them if desired.

- *Filter Type buttons:* Switch the noise generator's integrated filter between lowpass, highpass, and bandpass filter types.

  - *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. When set to LP, the filter operates as a lowpass filter. The filter slope is fixed at 12 dB/octave in LP mode.

  - *HP (highpass):* This filter type allows frequencies above the cutoff frequency to pass. When set to HP, the filter operates as a highpass filter. The filter slope is fixed at 12 dB/octave in HP mode.

- *BP (bandpass):* The frequency band directly surrounding the center frequency (determined with the Cutoff knob) is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 dB/octave on each side of the center frequency of the band.
- *byp (bypass):* Disables the integrated filter.

- *Cutoff and Resonance knobs:* Determine the cutoff/center frequency and resonance/bandwidth behavior of the integrated filter.
  - Cutoff can be modulated by the sources found in the “mod” and “via” menus.
  - The Cut knob defines the point in the frequency spectrum where the signal is boosted or cut. Depending on the selected filter type, you can make a sound darker (LP), thinner (HP) or more nasal (BP) by adjusting the Cut value.
  - Increasing resonance boosts frequencies that surround the cutoff frequency. Values range from 0 (no increase) to self-oscillation of the filter at high resonance values.

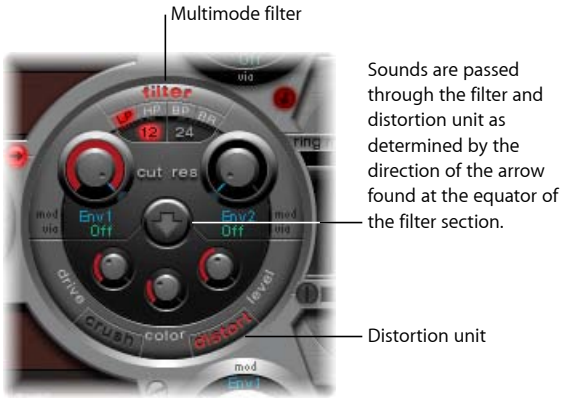
  Note: Self-oscillation is typical of analog filter circuits. It occurs when the filter feeds back into itself and begins to oscillate at its natural frequency, when high resonance values are used.
- *Dirt knob:* This parameter was especially developed for the noise generator. Higher values roughen up the pure, white noise appreciably, making it more grainy.
  - The Dirt parameter is especially effective at high resonance values.
  - Dirt can be modulated by the sources found in the “mod” and “via” menus.
- *Volume knob:* Sets the output level of the noise generator. Volume can be modulated by the sources found in the “mod” and “via” menus.
- *Signal Flow button:* Determines whether the noise generator signal is routed through the (main) filter, or directly to the EQ (output) section. When active, the button turns red and an arrow indicates the direction of the signal flow.

  Note: The signal flow button has no effect on the independent filter contained in the noise generator. The latter is deactivated with the byp button in the Type menu. It is therefore possible to filter the noise generator signal twice. In many instances you may want the noise generator signal to bypass the main filter, freeing the main filter for other duties—an important element when programming drum sounds.

Getting to Know the Ultrabeat Filter and Distortion Section

The output signals of both oscillators, the ring modulator, and the noise generator are passed on to Ultrabeat’s central Filter section (if not bypassed with the various signal flow buttons). The Filter section offers a multimode filter and a distortion unit.

For more information about synthesizer filters, see Filters.

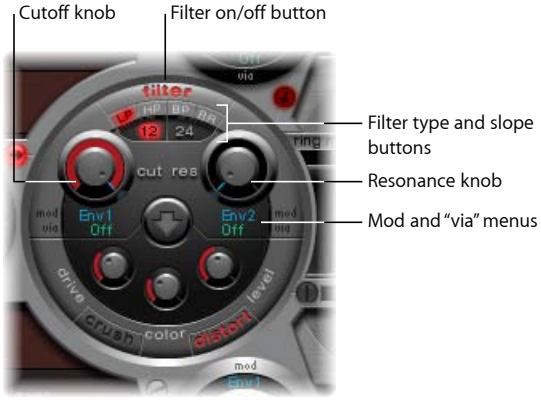

Multimode filter
Sounds are passed through the filter and distortion unit as determined by the direction of the arrow found at the equator of the filter section.
Distortion unit

To set the signal flow order through the filter and distortion unit

- Click the arrow to change the signal flow order between the following:
  - First the Distortion unit, then the Filter circuit (arrow pointing up)
  - First the Filter circuit, then the Distortion unit (arrow pointing down).

Using Ultrabeat's Multimode Filter

Ultrabeat features a powerful multimode filter that can dramatically, or subtly, alter the timbre of your drum sounds.

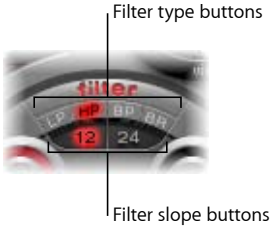


- *Filter (On/Off) button:* Activates or deactivates the entire Filter section. Deactivating the Filter section makes it easier to hear adjustments to other sound parameters, as the filter always heavily affects the sound. If the Filter label is red, the filter is engaged. If gray, the filter is disabled.

- *Filter Type buttons:* Switch the Filter between lowpass, highpass, bandpass or band-rejection filter types. See Choosing the Ultrabeat Filter Type (LP, HP, BP, BR).
- *Filter Slope buttons (12 and 24):* These buttons switch the filter between different slopes. See Setting the Ultrabeat Filter Slope,
- *Cutoff and Resonance knobs:* Determine the cutoff/center frequency and resonance/bandwidth of the filter. See Using Ultrabeat's Filter Cutoff Parameter and Using Ultrabeat's Filter Resonance Parameter.
- *"Mod" and "via" menus:* Determine the modulation source (and via source) for the Cutoff and Resonance parameters. See Working with Modulation in Ultrabeat.

Choosing the Ultrabeat Filter Type (LP, HP, BP, BR)

Ultrabeat's filter can operate in several modes, allowing specific frequency bands to be filtered (cut away) or emphasized.



Select one of the following buttons to choose a filter type:

- *LP (lowpass):* This filter type allows frequencies that fall below the cutoff frequency to pass. When set to LP, the filter operates as a lowpass filter. The slope of the filter can be set to 12 or 24 dB/octave in LP mode.
- *HP (highpass):* This filter type allows frequencies above the cutoff frequency to pass. When set to HP, the filter operates as a highpass filter. The slope of the filter can be set to 12 or 24 dB/octave in HP mode.
- *BP (bandpass):* The frequency band directly surrounding the center frequency (set with the Cutoff knob) is allowed to pass. All other frequencies are cut. The Resonance parameter controls the width of the frequency band. The bandpass filter is a two-pole filter with a slope of 6 or 12 dB/octave on each side of the center frequency of the band.
- *BR (band-rejection):* The frequency band directly surrounding the center frequency (set with the Cutoff knob) is rejected, while the frequencies outside this band can pass. The Resonance parameter controls the width of the rejected frequency band.

Setting the Ultrabeat Filter Slope

Most filters don't completely suppress the portion of the signal that falls outside the frequency range defined by the Cutoff parameter. Frequencies that are located close to the cutoff frequency are generally reduced less than those that are farther away. The higher the slope value, the more apparent the level difference is between frequencies that are near to the Cutoff frequency and those that are farther away from it.

The slope (curve) chosen for the Filter expresses the amount of rejection in decibels per octave. The Filter offers two different slopes: 12 dB and 24 dB per octave. The steeper the slope, the more severely the level of signals below the cutoff frequency are affected in each octave.

Using Ultrabeat's Filter Cutoff Parameter

The Cutoff Frequency (Cut) parameter controls the brilliance or determines the center frequency of the signal.

- In a lowpass filter, the higher the cutoff frequency is set, the higher the frequencies of signals that are allowed to pass.
- In a highpass filter, the cutoff frequency determines the point where lower frequencies are suppressed, with only upper frequencies allowed to pass.
- In a bandpass/band-rejection filter, the cutoff frequency determines the center frequency for the bandpass or band-rejection filter.

Using Ultrabeat's Filter Resonance Parameter

The Resonance (Res) parameter emphasizes or suppresses portions of the signal above or below the defined cutoff frequency, or determines the width of the band that surrounds the cutoff frequency.

- In a lowpass filter, Resonance emphasizes or suppresses signals below the cutoff frequency.
- In a highpass filter, Resonance emphasizes or suppresses signals above the cutoff frequency.
- In bandpass/band-rejection filters, resonance determines the width of the frequency band that surrounds the center frequency (set with the Cutoff Frequency parameter).

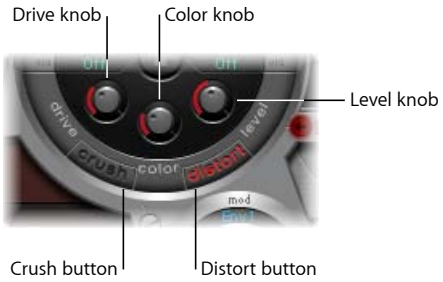
Using Ultrabeat's Distortion Circuit

The Distortion unit provides either a bit crusher or distortion effect.

The distortion effect is modeled on an analog distortion unit, which distorts the sound by overdriving the level. The bit crusher uses a process that deliberately reduces the digital resolution of the sound, measured in bits, achieving an intentionally digital coloration of the sound.

Both methods lead to distortions that are as tonally divergent as the two approaches. Distortion offers a more analog feel while the bit crusher can't hide its digital origins, nor is it supposed to.

Note: The arrow in the Filter section determines whether the Distortion circuit is inserted before or after the multimode filter (see Getting to Know the Ultrabeat Filter and Distortion Section).



- *Crush and Distort buttons:* Activate the desired mode. The name of the active effect type is indicated in red. If neither button is active, the Distortion circuit is bypassed.
- *Drive knob:* Controls the amount of distortion.
- *Color knob:* Determines the basic tone of the distortion. Higher values result in a brighter sound. Lower values lead to a darker, warmer tone.
- *Level/Clip knob:* Sets the output level of the Distortion effect when in Distortion mode. In Bit Crusher mode, this knob sets a threshold level for incoming signals from the sound sources that must be reached before distortion (bit crushing) begins.

Using the Ultrabeat Output Section

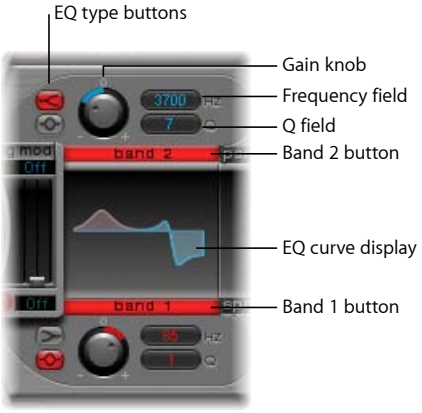
Depending on the status of each signal flow button, the output signals of both oscillators, the ring modulator, and the noise generator are routed to the Output section of Ultrabeat. This routing is either direct or through the Filter and Distortion section.

The Output section passes signals through both equalizers (EQ), then on to the Pan Modulation/Stereo Spread section (in a preconfigured order) before the final level is set for the selected sound and the trigger behavior is adjusted.

- *Two-Band EQ:* Provides precise tonal control of each drum sound.
- *Pan Modulation and Stereo Spread parameters:* Pan Modulation varies the panorama position of a drum sound. Stereo Spread broadens the stereo image.
- *Voice Volume:* Sets the default level of each drum sound.
- *Trigger Mode controls:* Determine the way that Ultrabeat reacts to incoming MIDI notes. This is independently defined for each sound.

Using Ultrabeat's Two-Band EQ

Both equalizer bands have almost identical features. Their parameters are explained jointly, but you can, of course, adjust band 1 (the lower EQ in the Output section) and band 2 separately.



- *Band 1 and Band 2 buttons:* Turn the individual band on or off. When active, the label is red. If neither EQ is activated, the signal passes through unaffected.
  - Band 1 offers a low shelving EQ whereas band 2 features a high shelving EQ.
- *EQ type buttons:* Switch between two different types of EQs: shelving and peak.
  - In shelving mode, activated by clicking the upper EQ type button, all frequencies above or below the set frequency are either increased or reduced.
  - In peak mode, activated by clicking the lower button, only frequencies located near the set frequency are affected.

  Note: Shelving EQs behave much like synthesizer lowpass and highpass filters. The key difference is that lowpass and highpass filters merely dampen certain frequencies (filter them out), whereas shelving EQs also allow these frequencies to be boosted.
- *Gain knob:* Positive values boost a certain frequency range as determined by the EQ type and Hz settings. Negative gain values reduce the gain of the frequency range. If the Gain knob is set to a value of 0, the EQ has no effect.
  - Option-click a Gain knob to set it to a neutral position. Alternatively, you can click the tiny 0 above the Gain knob.

- *Frequency (Hz) field:* Determines the frequency range to be boosted or reduced, by dragging vertically on the Hz value field.
  - Option-click the Hz parameter to set the value to a neutral position. This is 200 Hz for the first band and 2000 Hz for the second. The selection of these default frequencies was made in accordance with the different shelving characteristics of each frequency band. Band 1 is designed to filter low frequencies and band 2 is designed to filter high frequencies.
- *Q field:* Sets the Q (Quality) factor, by dragging vertically on the Q value field. The effect of Q on the sound is heavily dependent on the selected EQ type:
  - With shelving filters, as the Q value goes up, the area around the threshold frequency becomes more pronounced.
  - With the peak EQ, Q determines the width of the frequency band selection: low Q values select a broad band whereas high Q values select a very narrow band to be boosted or reduced with the Gain control.

Each EQ band displays parameter changes on a frequency response curve. The display provides immediate access to the Gain, Hz, and Q parameters of each band.

To directly edit the graphical EQ curve:

- Horizontally drag to change the EQ frequency.
- Vertically drag to change the Gain.
- Drag the handle shown at the peak (maximum point) of the EQ curve to change the Q factor.

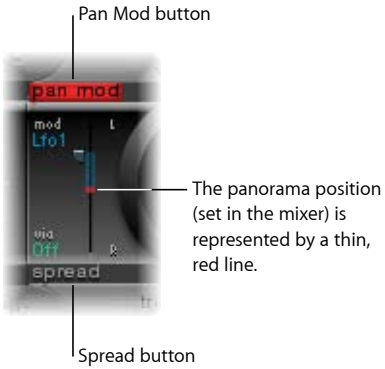
Using Pan Modulation and Stereo Spread in Ultrabeat

The EQ's output signal is passed along to the Pan Modulation/Stereo Spread section where the placement of the sound in the stereo field can be modulated (Pan Modulation mode), or the stereo basis of the sound can be broadened (Stereo Spread mode).

Ultrabeat Pan Modulation Mode

Pan Modulation varies the panorama position of a drum sound with a "mod" (and "via") source.

Note: The modulation set here is relative to the panorama position set in Ultrabeat's Mixer in the Assignment section.



- *Pan Mod and Spread buttons:* Activate the corresponding mode. If neither mode is activated, the signal passes through unaffected.
- *Mod and via menus:* Determine the modulation and via sources for pan modulation.
- *Mod and via sliders:* The blue control on the left and the green control on the right are used to set the amount (intensity) of mod and via modulation.

  Note: You cannot directly move the red line that represents pan position shown in this section. In order to move the line, rotate the pan knob in the Mixer section.

Ultrabeat Stereo Spread Mode

Stereo Spread broadens the stereo image, making it wider and more spacious.

- *Lo Freq(uency) slider:* Adjusts the width (the spreading effect) of bass frequencies: the higher the value, the more prominent the effect becomes.
- *Hi Freq(uency) slider:* Adjusts the width of high frequencies.

Controlling Ultrabeat Drum Sound Levels with Voice Volume

The Voice Volume knob adjusts the output volume of individual drum sounds. To be more exact, you are setting the maximum level for the selected drum sound with Env 4, following the attack phase of Env 4.

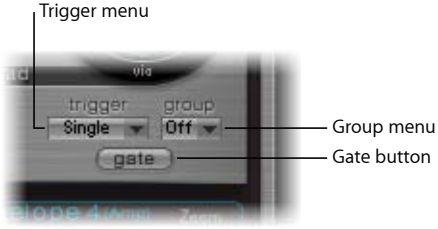
Note: Envelope 4 (Env 4) is hard-wired to voice volume—the level control for the selected sound. Each sound in the kit also has a further three envelopes and other modulation sources available for control of other synthesis parameters.

The intensity of Envelope 4's impact on Voice Volume can also be modulated with a via source.

Note: Voice Volume precedes the sliders in the mixer. This approach allows the starting volume of individual drum sounds to be set independently of their relative levels in the drum kit mix, which you adjust in the Assignment section mixer.

Changing the Ultrabeat Trigger Mode

The way Ultrabeat reacts to a succession of incoming notes is independently defined for each sound. Parameters that provide control over this aspect of Ultrabeat's behavior are found in the Trigger Mode section, below the Voice Volume knob.



- *Trigger pop-up menu:* Enables you to choose between Single and Multi trigger modes.
  - *Single:* A new trigger note cuts off the note that is currently playing.
  - *Multi:* When a new note is played, currently playing notes continue to decay in accordance with their respective amplitude envelope settings (Env 4).
- *Group pop-up menu:* Enables you to choose between Off and settings 1 through 8. If two different sounds are assigned to the same group, they will cut each other off when a new note message is received. Only one sound in the group can be played at a time.
  - A typical use of this feature is when you're programming hi-hat sounds: when playing a real hi-hat, the closed hi-hat note cuts off or mutes the ringing of the open hi-hat. This feature is often referred to as hi-hat mode.

Note: While in Single Trigger mode, only the currently sounding note of the same sound is cut off. A sound that is assigned to a group cuts off all other sounds, regardless of note, in the group.

- *Gate button:* Enables or disables the Gate function. When active, the sound is immediately cut off when the MIDI note is released (MIDI note off), regardless of envelope settings.

  Note: The Gate function ensures that a specific sound does not play—it can't be heard—after a note-off event, as defined in a host sequencer such as Logic Express, or Ultrabeat's internal sequencer. Note length can be an important creative element when you are programming rhythm tracks.

Working with Modulation in Ultrabeat

Most sound parameters can be controlled dynamically (modulated) in Ultrabeat. Ultrabeat provides two LFOs, four envelope generators, velocity, and four freely definable MIDI controllers as modulation sources. The setting of modulation routings follows a universal principle that works as follows:

Ultrabeat's modulation routings feature three key elements:

- *The modulation target:* The synthesizer parameter that you want to modulate.
- *The modulation source:* The parameter that modulates the target.
- *The via source:* A secondary modulation source that can influence the intensity of the first modulation source.

Note: You can freely use the same sources and the same via controllers in multiple modulation routings.

Mod and Via Modulations in Ultrabeat

You can modulate a sound parameter using an adjustable value—the modulation depth or intensity—with the "mod" parameter. You can choose between two LFOs, four envelope generators, and Max as sources for this modulation.

Via allows you to further tailor the modulation effect. The depth of the first modulation ("mod") can be modulated by a separate, independent source. The intensity of this effect is set with the "via" parameter. The sources for via modulations include velocity and four freely definable MIDI controllers.

A typical via modulation usage example would be increasing a pitch sweep as you play at higher velocities. To this end, you would use:

- An envelope (Env) as the "mod" source for the pitch of an oscillator
- Velocity (Vel) as the via source

The harder the key is played, the higher (in pitch) it will sound—which is ideal for synthesized tom-tom sounds, for example.

Ultrabeat Modulation Examples

Consider the following example to better understand how this works:

The default Cut (Cutoff) parameter value is 0.50. No modulation source has been selected in either the blue "mod" or green "via" menu (both are Off) in the image below.

When a modulation source is selected in the "mod" menu (Env 1 in the image below), the ring around the rotary knob is activated. Drag in the ring to set the desired value (0.70 in the example) for the Cut parameter—when affected by the "mod" source.

Note: Exact values are shown in the help tags when adjusting various parameters.

As soon as a modulation source is selected in the "via" menu (Ctrl A in the image below), a movable slider appears on the "mod" ring. Drag this slider to set the maximum modulation value that can be reached through use of the via source (0.90 in this example).

So much for the settings. What do the markings found around the Cut knob mean, and what is happening to the sound?

The "mod" and "via" controls indicate the minimum and maximum values that the modulated parameter can attain, in comparison to the default value.

In the example, the Cut(off) frequency of the filter is set to a default value of 0.50. The "mod" source (Env 1) drives the Cut value up from 0.50 to 0.70 during the attack phase and back down to 0.50 during the decay phase.

When the via source (Ctrl A) is introduced, the following interplay occurs: when Ctrl A is at its minimum value, nothing changes; Cutoff continues to be modulated between values of 0.50 and 0.70 by the envelope (Env 1). A maximum value for Ctrl A causes the envelope generator to vary the parameter between the values of 0.50 (the default Cut value) and 0.90 (the via amount).

You can see at a glance the degree of maximum influence on basic parameters by the mod and via modulation sources—the area between the mod and via points shows the amount that the modulation depth can be further altered by the via modulation source. In the example, the Cutoff can reach values between 0.70 and 0.90 depending on the value being sent by Ctrl A.

Here's another example:

Cutoff is again set to 0.50, Env 1 now drives the value down to 0.25, and a maximum Ctrl A value reduces the Cutoff frequency down to 0.

The example below illustrates the simplicity and speed of Ultrabeat's modulation options:

In this example, the modulation intensity of Env 1, which affects Cutoff, is controlled with the dynamics of the performance (Vel). The secondary via modulation also controls its direction. Try this setting in Ultrabeat to create some extremely interesting sounds.

Creating an Ultrabeat Modulation Routing

The following applies to all parameters that offer "mod" (and "via") modulation options.

To create a modulation routing

1 Click the "mod" label of the desired parameter to open the "mod" pop-up menu.

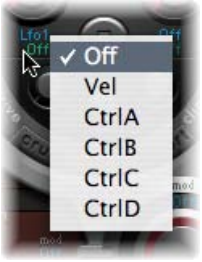
2 Choose one of the modulation source settings in the menu:

- The Off setting deactivates the “mod” routing, and the “mod” control can no longer be adjusted. In this situation, no “via” modulation can occur either, because “via” no longer has a modulation target, and the “via” control disappears.
- The Env settings set one of the envelope generators as modulation source
- The Max setting produces a static modulation at maximum level. When the “mod” value is set to Max, the “via” parameter is routed directly to the modulation target. This way, velocity can be used as a direct modulation source, even though Vel is not available as a source in the “mod” menu.

  Tip: You can also set up an external MIDI fader unit with Ctrl A, B, C, or D (see Using Ultrabeat MIDI Controllers A-D). You can then use the Max menu item to route the respective via source—Ctrl A, B, C, or D—to the parameter you’d like to control with one of the faders on your MIDI fader box.

3 If you want to assign a via source, click “via” to open the “via” pop-up menu.

4 Choose Vel or one of the parameters from Ctrl A through Ctrl D.



- Vel represents velocity.
- Ctrl A to Ctrl D are four continuous controllers that can be assigned to four external MIDI controllers. These assignments are made in the MIDI Controller Assignments area at the upper-right edge of the Ultrabeat window. The assignments apply to all sounds in the current Ultrabeat plug-in instance.

5 Adjust the “mod” and “via” controls as desired.

Using Ultrabeat MIDI Controllers A-D

The MIDI Controller Assignments area at the upper edge of the Ultrabeat window enables you to assign any MIDI controller shown in the menus to each of the four controller slots: Ctrl A, B, C, or D.

These assignments enable external MIDI controller hardware—such as sliders, knobs, aftertouch, or the modulation wheel of your MIDI keyboard—to control via modulation sources in Ultrabeat.

To assign a controller

- Open the desired control menu (Ctrl A–D), and choose the controller name or number that you want to use from the list.

To learn a controller assignment via MIDI

1 Open the desired control menu, and choose the -Learn- menu item.

2 Move the desired controller on your MIDI keyboard or controller.

Note: If no suitable MIDI message is received within 20 seconds, the selected control reverts to the previous value/assignment.

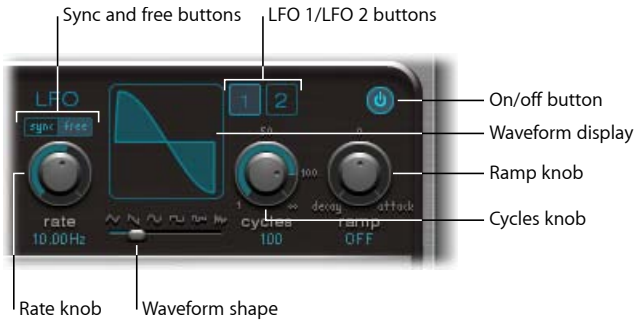
Getting to Know the Ultrabeat LFOs

Two identical LFOs are available as modulation sources in the “mod” menu.

LFO is the abbreviation for *low frequency oscillator*. The LFO signal is used as a modulation source. In an analog synthesizer, the LFO frequency generally ranges between 0.1 and 20 Hz, which is outside the audible frequency spectrum. Therefore, this type of oscillator is used only for modulation.

Note: The speed of the LFO in Ultrabeat can reach up to 100 Hz, which affords a number of possibilities that analog synthesizers simply don’t offer.

The parameters for both Ultrabeat LFOs are described below. You can freely adjust LFO 1 and LFO 2 independently of each other.



- *LFO 1 and 2 buttons:* Select the corresponding LFO, allowing independent parameter adjustments for each.
- *On/Off button:* Activates or deactivates the selected LFO.
- *Sync/Free buttons:* The LFO speed (Rate) can be set independently (Free) or synchronized (Sync) to the host application tempo. Click either button to activate the corresponding mode.
- *Rate knob:* Determines the speed of the LFO. Depending on the Free/Sync setting, the rate is displayed in Hertz or rhythmic values—the latter when project tempo synchronization is active. Rates range from speeds of 1/64 notes through to a periodic duration of 32 bars. Triplet and punctuated values are also available.
- *Waveform Shape slider (and display):* Determines the shape of the LFO waveform. Drag the slider from left to right to morph the waveform from a triangle, to a sawtooth, sine, square, and finally a rectangular wave shape—including all variations in-between. At the far right position, the LFO produces random waveforms. The graphical display shows the current LFO waveform shape.
- *Cycles knob and field:* Determines the number of times that the LFO waveform runs through. See Using Ultrabeat's LFO Waveform Cycles Parameter.
- *Ramp knob and field:* Controls the time it takes for the LFO modulation to fade in or fade out. The Ramp value is displayed in milliseconds in its parameter field.
  - Turn Ramp to the right to set the Attack time of the LFO.
  - Turn Ramp to the left to set the Decay time.
  - At the middle position, Ramp has no effect on the LFO.

Using Ultrabeat's LFO Waveforms

The Waveform Shape slider allows you to choose different waveforms for both LFOs. The table below outlines how these can affect your sounds. In-between waveform shapes will result in hybrid waveforms and hybrid behaviors.

Waveform	Comments
Triangle	Well suited for vibrato effects
Sawtooth	Well suited for helicopter and space gun sounds. Intense modulations of oscillator pitch with a sawtooth wave lead to "bubbling" sounds. Intense sawtooth modulations of lowpass filter cutoff and resonance create rhythmic effects.
Sine	Ideal for smooth, even modulations. Its position on the Waveform Shape slider enables you to smoothly morph between sawtooth and square/rectangular waves.
Square and Rectangle	Use of the square/rectangular waves will periodically switch the LFO between two values. The right-hand rectangular wave switches between a positive value and zero. The left-hand rectangular wave switches between a positive and a negative value set to the same amount above/below zero.
Sample & Hold	The right-hand waveform on the Waveform Shape slider outputs*random* values. A random value is selected at regular intervals, as defined by the LFO rate. The term*Sample & Hold* (S & H) refers to the procedure of taking samples from a noise signal at regular intervals. The values of these samples are then*held* until the next*sample* is taken. *Tip:* A random modulation of oscillator pitch leads to an effect commonly referred to as a*random pitch pattern generator* or*sample and hold*. Try using very high notes, at very high rates and high intensities—you'll recognize this well-known effect from hundreds of science fiction movies!

Using Ultrabeat's LFO Waveform Cycles Parameter

An LFO normally oscillates continuously. On percussive signals it can, however, be interesting to limit the LFO cycles (repetitions of the entire waveform) to a defined number. Ultrabeat allows you to set the number of LFO cycles with the Cycles parameter. After completing the defined number of cycles, the LFO stops oscillating.

Tip: Try small Cycles parameter values, with the LFO source used to control the Volume (Level) of one or both oscillators. This will create typical drum flams or hand claps.

The range of Cycles parameter values extends from 1 to 100. Turn the knob to its maximum value (all the way to the right) to create an infinite number of cycles (standard LFO behavior). A Cycle value of 1 allows the LFO to function as an additional, albeit simple, envelope generator.

The Cycles parameter can also determine whether the LFO (waveform) is started from the beginning, at a 0-crossing point, with each note trigger, or whether it simply continues oscillating.

A Cycles value of Inf (Infinity) forces the LFO to run freely. It is not reset by incoming MIDI note on messages.

When Cycles is set to values under 100, the LFO will be reset by each new MIDI note on message (Note On Reset).

It's really a question of taste as to whether or not you choose to trigger an LFO cycle from the same spot or just allow it to oscillate freely, regardless of phase. The random element of free-running LFOs can make many sounds fatter. This, however, can be at the expense of a percussive attack—which is not ideal for many drum sounds.

Note: You can, of course, use minor shifting of the LFO phase—with the Cycle value set to Infinity—to your advantage, adding an analog character to a drum sound, for example.

Getting to Know the Ultrabeat Envelopes (Env 1 to Env 4)

Further modulation sources available to you in the "mod" menu include the envelope generators.

Note: The roots of the term "envelope generator" and its basic functionality are described in The Attack, Decay, Sustain, and Release (ADSR) Envelope Controls.

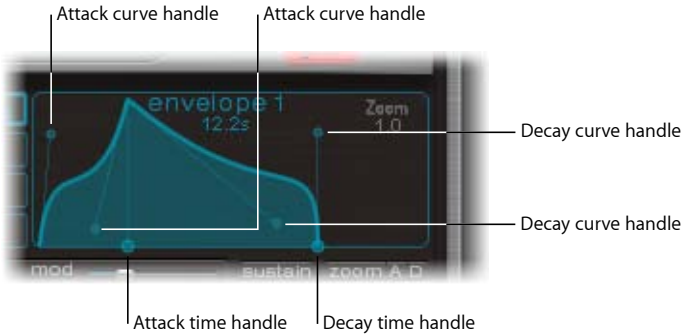
Ultrabeat features four identically specified envelope generators per voice. They are abbreviated as Env 1 to Env 4. In addition to potential use as a modulation source (in the "mod" menus of various sound parameters), Env 4 is permanently connected to the Voice Volume parameter. In other words, each Ultrabeat drum sound has a hard-wired volume envelope generator—namely, Env 4.

The default behavior of the envelope generators is known as the*one shot envelope mode:* after a key is pressed (note on message), the envelopes run their course, regardless of how long the note is held. This setting is ideal for percussive signals because it emulates the natural behavior of acoustic percussion instruments.

For special cases, such as sustained pad or cymbal sounds, you can activate a sustain mode where the envelopes take the lengths of the played notes into account.

Editing Ultrabeat's Envelopes Graphically

Ultrabeat's envelope display provides a unique envelope design, consisting of Bezier curves in which two segments—attack and decay—constitute the entire envelope.



In the envelope graphic, you can see various handles (junction points) of two different sizes.

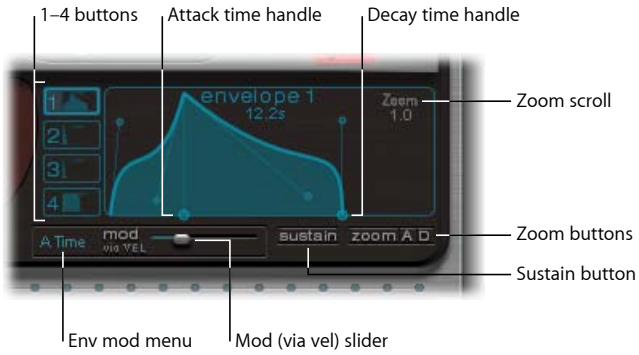
Both of the larger handles on the x-axis (the horizontal, or time axis at the bottom) control the attack and decay times, respectively. A vertical line extends up from the first of the two handles (attack), and divides the envelope into an attack and decay phase.

Both segments each have two small curve handles. You can drag these in any direction to deform the contour of the envelope and freely shape its amplitude.

You can also directly drag anywhere on the curve itself to reshape the envelope.

Using Ultrabeat's Envelope Parameters

In order to edit envelope parameters, you first need to select one of the four envelopes with buttons 1 through 4. After this is done, the parameters of the corresponding envelope can be changed in the envelope display window.



- *Buttons 1–4:* Used to choose one of the four envelopes. Only the selected envelope can be edited. The button frame of the chosen envelope is highlighted, and the envelope display immediately updates to reflect your selection.
- *Attack Time handle:* Defines the period of time the envelope needs to reach its maximum value after receiving a note on message. This period is called the*attack phase*.
- *Decay Time handle:* Defines the period of time the envelope needs to fall back to a 0 amplitude, after it has reached its maximum value (defined in the attack phase).

  Note: You can freely change the time and shape of the envelope in the attack or decay phase. See Editing Ultrabeat's Envelopes Graphically.
- *Zoom scroll field:* Resizes the visible contents of the envelope display when you drag horizontally.
- *Env "mod" menu:* Determines the modulation target (either the time or shape of the envelope attack or decay phase) by velocity. Choices are A Time, A Shape, D Time, or D Shape.
- *"Mod" (via vel) slider:* Determines the intensity of velocity modulation of the target specified in the Env "mod" menu.
  - When you modulate Shape, low velocity values make the envelope sag. Higher values make the envelope bulge.
  - When you modulate Time, high velocity values reduce the length of the envelope segment. Lower velocity values increase the length of the envelope segment.
- *Sustain button:* When active, displays a red handle (and vertical line) on the x-axis. This can be moved horizontally—but only within the envelope decay phase. The amplitude that the envelope reaches at the Sustain junction point is retained until the MIDI note is released.

Note: If the Sustain button is not activated, the envelope functions in one-shot mode, and the note length (MIDI note-off command) is disregarded.

- *Zoom (to fit) button:* Enlarges the envelope to fill the entire width of the envelope display, making it easier to adjust junction points and curves.

  Note: When the Zoom function is active, the decay handle can be dragged beyond the right-hand edge of the envelope display area, in order to lengthen the decay time. After you release the mouse button, the envelope graphic is automatically resized to fit the display area.

- *Zoom A/D buttons:* Shows only the attack (A) or decay (D) phase across the entire width of the envelope display. This facilitates more accurate edits to envelope shapes, even down to millisecond values.

Modulation Target Display of Ultrabeat LFOs and Envelopes

Ultrabeat's user interface has a feature that makes finding the modulation targets of the LFOs and envelopes faster: Simply click the numerical field of the desired modulation source to highlight all of its modulation targets.

Working with the Ultrabeat Step Sequencer

Ultrabeat also incorporates a powerful, integrated step sequencer, which you can use to create polyphonic rhythmic sequences and patterns. The sequencer displays running light-style controls like those of classic drum machines and shares many of the sequence and pattern creation methods employed in these devices.

Ultrabeat's sequencer expands on the features of hardware drum machines by providing extensive automation and editing features. These enable you to precisely vary the timbre of the sound and the overall dynamics—at any point in the pattern. The sequencer plays an important role in shaping the rhythms and sounds that you can produce with Ultrabeat.

The step sequencer allows all Ultrabeat sounds to be combined in patterns, based on sequences for each individual sound. Its design and use—commonly referred to as step programming—are based on analog sequencers and drum machines. Unlike these analog precursors, Ultrabeat enables you to program automated changes for nearly every synthesizer parameter.

Depending on your personal taste and favored musical style, you'll want to control Ultrabeat from either the integrated step sequencer or from Logic Express when programming rhythms. Combining both sequencers is also possible; they can be active at the same time and are automatically synchronized with each other. Logic Express acts as a master clock in this situation, determining the tempo of Ultrabeat's internal step sequencer.

If you're unfamiliar with the concept of step sequencing, the following section will help you understand the design of Ultrabeat's step sequencer.

How Step Sequencers Work

The basic idea behind analog step sequencers is to set up a progression of control voltages and output these step-by-step, typically in an endlessly repeating pattern. This basic principle helped to spawn a number of electronic music styles based on the mesmerizing effect that repeating patterns can have.

In early analog sequencers, three control voltages were usually created per step in order to drive different parameters. The most common usage was control of a sound's pitch, amplitude, and timbre (cutoff) per step.

The control surface of analog sequencers often contained three rows of knobs or switches aligned on top of (or beside) each other. Each row commonly contained 8 or 16 steps. Each row provided a control voltage output that was connected to a control input (for a particular parameter) on a synthesizer. A trigger pulse determined the tempo between steps. A running light (an LED) indicated the step that was currently being triggered.

The running light programming concept also appeared in later drum computers, the most well-known examples being the Roland TR series drum machines.

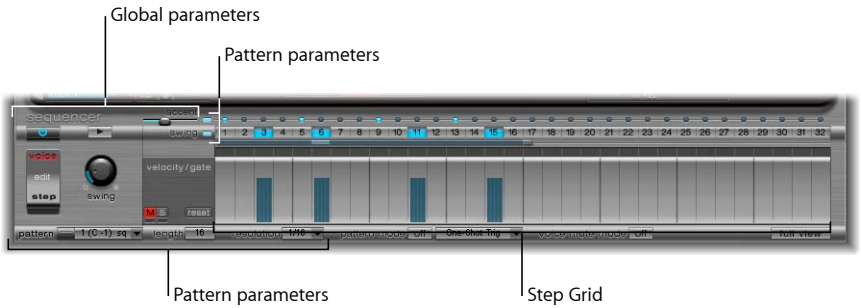
The introduction of the MIDI standard and increased use of personal computers for music led to a rapid decline in the step sequencer and related technology. More flexible recording and arranging concepts that didn't adhere to the step and pattern principle were possible with the far more powerful personal computer.

Despite these technological advances, step sequencers haven't disappeared completely. Hardware groove boxes have experienced a renaissance in recent years due to their intuitive nature, which has made them a favored tool for rhythm programming.

Ultrabeat's integrated step sequencer couples the advantages and general working principles of its analog forebears with far more flexible control options, raising modern rhythm programming to a new level.

Getting to Know Ultrabeat's Step Sequencer

Ultrabeat's step sequencer contains a sequence for each sound in a drum kit. Each sequence can consist of up to 32 steps. A pattern is a container for all sequences in a drum kit. Up to 24 patterns can be saved and recalled with each Ultrabeat setting.



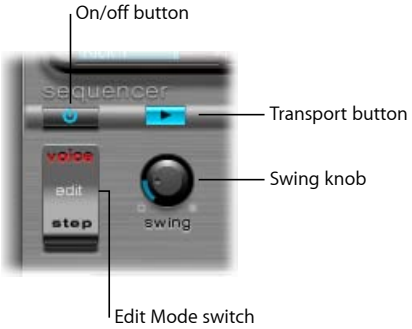
The sequencer is divided into three sections.

- *Global parameters:* Enable the step sequencer, control playback, provide access to various modes, and control the overall playback feel.
- *Pattern parameters:* Provide control over the length and resolution of the currently selected pattern. You can also accentuate individual steps in the pattern, for each drum sound.
- *Step Grid:* This is where actual sequencing takes place. A sequence of up to 32 steps, for the sound that is currently selected in the Assignment section, is shown. You can add, remove, or alter events in the grid.

Note: An alternate view allows you to simultaneously see and edit the steps of all drum sounds in the pattern. See Switching the Ultrabeat Step Grid to Full View.

Using Ultrabeat's Global Sequencer Parameters

A description of the parameters that apply globally to all patterns follows.

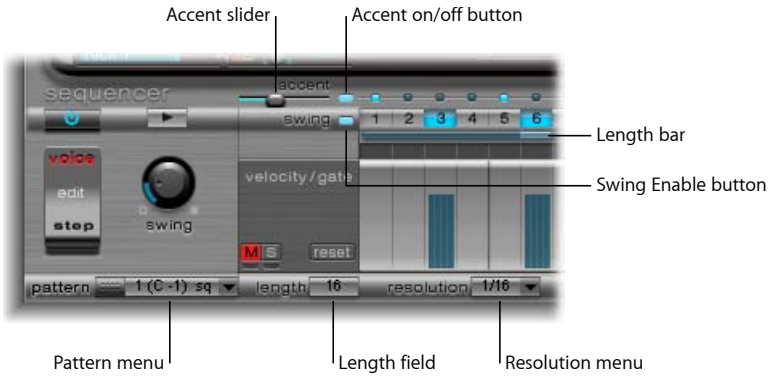


- *On/Off button:* Enables or disables the step sequencer.
- *Edit Mode switch:* Chooses either Voice or Step mode.
  - *Voice mode (default):* In Voice mode, editing a drum sound's parameters sets the parameters of the drum sound itself.
  - *Step mode:* In Step mode, you can automate a sound's parameters from one step to the next. For more information see Automating Parameter Values in Ultrabeat's Step Sequencer.
- *Transport button:* Starts and stops the sequencer pattern. The step sequencer is always synchronized to the tempo of the host application.

  Note: If the Transport button is displayed in blue, the sequencer interprets incoming MIDI notes between C-1 and B0 as performance information. For more information see Using MIDI to Control Ultrabeat's Sequencer.
- *Swing knob:* Determines the swing intensity for all sounds that have the Swing function activated (see Using Ultrabeat's Swing Function).

Using Ultrabeat's Pattern Parameters

A pattern contains all events, stored in sequences, for all 25 sounds. At the bottom edge of the Ultrabeat window you can select from one of 24 patterns, and set parameters that globally affect all sounds in the kit.



- *Pattern menu:* Allows you to choose one of the 24 patterns.
- *Length field and bar:* Defines the length of the pattern. You can adjust the length of the grid by dragging the value in the Length parameter field or the bar beneath the swing buttons.
- *Resolution menu:* Determines the resolution of the pattern. It defines the metric unit of a measure that is represented by the individual steps. For example, the 1/8 setting means that each step of the grid represents an eighth note. Given a pattern length of 32 steps, the pattern would run for 4 measures (the 32 setting applies to the entire grid and, therefore, equally to all sounds).

  Note: The interplay between the Length and Resolution values allows the creation of different kinds of time signatures. For example, the values Length = 14 and Resolution = 1/16 results in 7/8 time; Length = 12 and Resolution = 1/16 results in 3/4 time; and Length = 20 and Resolution = 1/16 results in 5/4 time.
- *Accent button and slider:* Allows individual steps to be strongly emphasized, or accentuated. See Using Accents in Ultrabeat.
- *Swing Enable button:* When enabled, the grid of the currently selected sound is played in accordance with the Swing knob setting. See Using Ultrabeat's Swing Function.

Copying and Reorganizing Ultrabeat Patterns

You can reorganize the 24 patterns in the Pattern menu, using Copy and Paste commands.

To copy a pattern using a shortcut menu

1. Choose the desired pattern from the Pattern menu.
2. Control-click (or right-click) the Pattern menu, then choose Copy from the shortcut menu.
3. Choose the target pattern from the Pattern menu.

4 Control-click the Pattern menu, then choose Paste from the shortcut menu.

You can also use a key command to copy patterns.

To copy a pattern using the key command

1 Select the desired pattern in the Pattern menu.

2 Press Option, open the Pattern menu and select another Ultrabeat pattern. This replaces the pattern in the target position.

Note: Be aware that all existing sequencer data in the target pattern will be replaced. If you change your mind during the process, select the source pattern number.

To clear a pattern

1 Select the desired pattern in the Pattern menu.

2 Control-click (or right-click) the Pattern menu, then choose Clear from the shortcut menu.

Using Ultrabeat's Swing Function

Swing changes the distance between notes. Only even-numbered steps are affected by the Swing parameter, with notes on odd-numbered steps remaining where they are.

Which beats are affected depends on the selected Resolution parameter setting, as illustrated in the following example: At a Resolution of 1/8 and a Length of 8, the notes on steps 1, 3, 5, and 7 represent quarter notes in the measure. These remain unchanged. Only the eighth notes found between them (steps 2, 4, and so on) are shifted by the Swing function. The amount of shift is equal to the swing intensity (set with the Swing knob).

Note: Swing is active only for grid resolutions of 1/8 and 1/16 (see Using Ultrabeat's Pattern Parameters).

To use the swing function

1 Click the Swing Enable button.

This forces the grid of the currently selected sound to be played in accordance with the Swing knob setting.

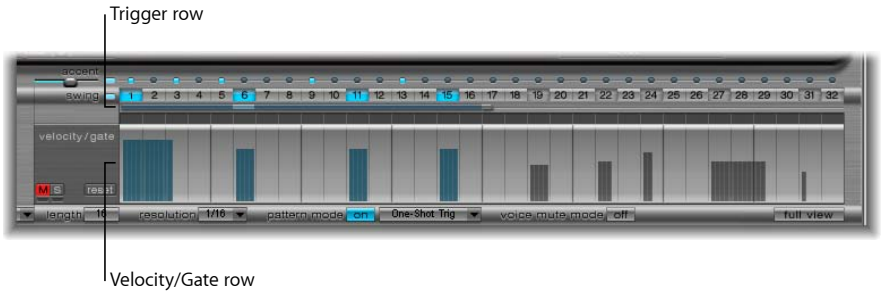
2 Adjust the Swing knob.

At a 0 setting, with the knob all the way to the left, the Swing function is disabled. Turn the knob to the right to move affected notes toward the following note.

Getting to Know the Ultrabeat Step Grid

The step grid displays sequence steps on two rows. The steps shown in these rows correspond to the sound that is currently selected in the Assignment area. Choosing a different sound switches the sequencer display to show the rows that correspond to the newly selected sound.

The step grid area contains two rows, each with 32 fields (steps).


Trigger row
accent
swing
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32
velocity/gate
M S
reset
length
resolution 1/16
pattern mode on
One-Shot Trig
voice mute mode off
full view
Velocity/Gate row

- *Trigger row:* Click a button to activate or deactivate the sound on the corresponding beat.
- *Velocity/Gate row:* Sets the length (gate time) and velocity of steps entered in the Trigger row. Both parameters are displayed as a single graphical bar. The bar's height represents the velocity; its length, from left to right, depicts the note length.

Creating and Removing Steps in the Ultrabeat Trigger Row

The Trigger row consists of buttons 1 to 32, which represent steps or beats in the selected sequence. Note trigger events are placed on corresponding steps. This is where you designate when (on which beat) the selected sound plays.

Note: You can create and remove steps while the step sequencer is running.

To create steps

1. Choose the desired sound in the Assignment section.
2. Enable the step sequencer by clicking the On button.
3. Choose a pattern and set the desired length and resolution. For details, see Using Ultrabeat's Pattern Parameters.
4. Click the desired buttons, 1 through 32, to activate or deactivate the selected sound on the corresponding beat. In the example shown above, these are steps 1 and 6.

   Note: An alternate view allows you to simultaneously see and edit the steps of all drum sounds in the pattern. See Switching the Ultrabeat Step Grid to Full View.

To remove a step

1. Ensure that the appropriate sound is selected in the Assignment section.
2. Click the desired buttons, 1 to 32, that correspond to the step(s) that you want to remove.

Note: Drag horizontally across the buttons to quickly enable or disable trigger events.

Using the Ultrabeat Trigger Shortcut Menu Commands

Control-click (or right-click) any of the trigger buttons to open the Trigger shortcut menu, which offers the following commands:

Copy, Paste, and Clear Commands

- *Copy:* Copies all activated triggers (steps) to the Clipboard.
- *Paste:* Pastes all triggers from the Clipboard.
- *Clear:* Turns off all activated triggers.

Create Beat Commands

- *Add Every Downbeat:* Adds triggers on every downbeat in the sequence. The exact determination of which steps are downbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Downbeat would create triggers on every 4th step. Starting with the initial downbeat at step 1, this would create trigger events on step 5, step 9, step 13, and so on. This command does not erase existing trigger events; it only adds trigger events.
- *Add Every Upbeat:* Adds triggers on every upbeat in the sequence. The exact determination of which steps are upbeats depends on the grid resolution. For example, if the resolution is set to 1/16, Add Every Upbeat would create triggers on every 4th step. Starting with the initial upbeat at step 3, this would create trigger events on step 7, step 11, step 15, and so on. This command doesn't erase existing trigger events; it only adds trigger events.

Alter, Reverse, and Shift Commands

- *Alter Existing Randomly:* Randomly reorders steps while retaining the number of activated triggers.
- *Reverse Existing:* Reverses the order of existing steps.
- *Shift Left by 1 Step:* Shifts all steps in the sequence one step to the left.
- *Shift Left by 1 Beat:* Shifts all steps in the sequence one beat to the left. The precise number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.
- *Shift Left by 1/2 Beat:* Shifts all steps in the sequence one-half beat to the left. The precise number of steps that equals one half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so half of one beat equals one step, and so on.
- *Shift Right by 1 Step:* Shifts all steps in the sequence one step to the right.

- *Shift Right by 1 Beat:* Shifts all steps in the sequence one beat to the right. The precise number of steps that equals a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps; at a resolution of 1/8, a beat equals two steps, and so on.
- *Shift Right by 1/2 Beat:* Shifts all steps in the sequence one-half beat to the left. The precise number of steps that equals one half of a beat depends on the current grid resolution. For example, at a resolution of 1/16, a beat equals four steps, so half of one beat is two steps; at a resolution of 1/8, a beat equals two steps, so one half of one beat is one step, and so on.

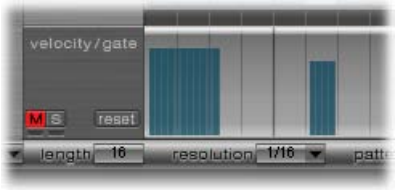
Create and Replace Commands

- *Create & Replace Randomly:* Erases, then randomly creates new, steps in the sequence; in other words, a brand new sequence is created. The number of events that are created depends on the grid resolution.
- *Create & Replace Few:* Similar to Create & Replace Randomly, but only a handful of new steps are created. The number of steps that are created depends on the grid resolution.
- *Create & Replace Some:* As above, but more new steps are created. The number of steps created depends on the grid resolution.
- *Create & Replace Many:* As above, but a large number of new steps are created, effectively filling the pattern.

  For example, start with an empty sequence of 32 steps at 1/16 resolution. Using Create & Replace Few creates 4 new steps; using Create & Replace Some creates 8 new steps; and using Create & Replace Many creates 16 new steps.

Setting Step Lengths and Velocities in Ultrabeat's Velocity/Gate Row

This row enables you to set the length (gate time) and velocity of the notes entered in the Trigger row. Both parameters are displayed as a single graphical bar. The bar's height represents note velocity, and its width indicates the note length (gate time).



To change the length and velocity values for each step

- Drag vertically on the blue bar to change the velocity.
- Drag horizontally on the blue bar to change the note length (gate time).

The gate time is divided into four equal sections, making it easy to set rhythmically accurate note lengths. In order for the one-shot envelope to react to gate time, it is necessary to either activate the Gate function in the sound itself (see Changing the Ultrabeat Trigger Mode) or use envelopes in sustain mode (see Using Ultrabeat's Envelope Parameters), in conjunction with rhythmically useful (short) decay times.

To reset all velocity and gate values to default settings

- Click the Reset button at the left of the velocity/gate row.

The default velocity setting = 75 percent. The default gate time = all four sections are active.

Using Ultrabeat's Velocity/Gate Shortcut Menu Commands

Control-click (or right-click) any step in the velocity/gate row to open a shortcut menu that has the following commands:

- *Alter Vel(ocities):* Changes the velocity values of all steps by a random amount, while retaining the selected beats (the Trigger row remains unchanged).
- *Alter Gate (Time):* Changes the note lengths of all steps by a random amount, while retaining the selected beats (the Trigger row remains unchanged).
- *Randomize Vel(ocities):* Creates a new, random velocity value.
- *Randomize Gate (Time):* Creates a new, random gate value.

Using Accents in Ultrabeat

The Accent setting can be switched on or off individually for each drum sound. This allows you to turn accents on for cymbals but turn accents off for the kick drum, for example.

To enable accents and set the accent level

1. Click the blue LED to the right of the Accent slider to enable the accent function.
2. Move the Accent slider to globally determine the volume of programmed accents.

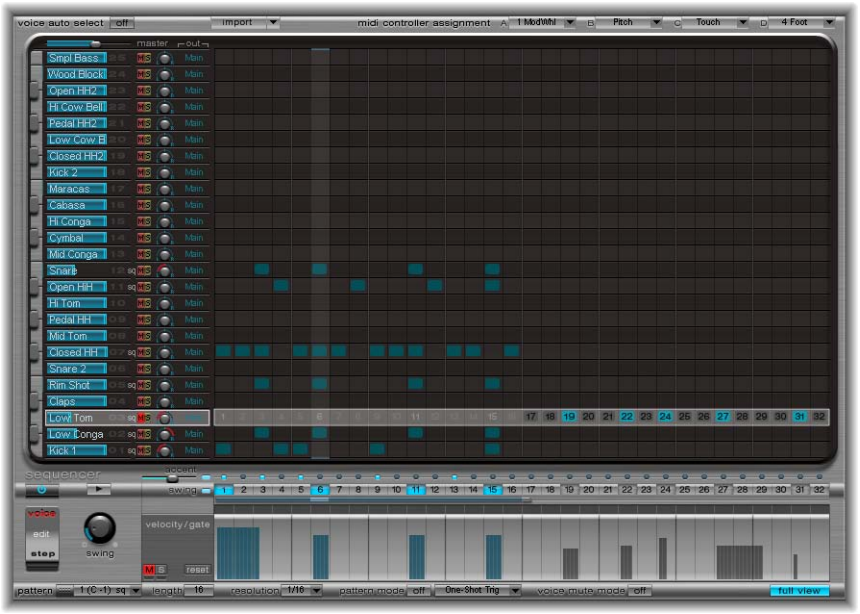
To program an accent for a particular step

- Click the blue LED above the desired step (steps 1 and 3 in the image). The sound at this step position will be accentuated (played louder).

Switching the Ultrabeat Step Grid to Full View

Click the Full View button in Ultrabeat's lower-right corner to see a large sequencer grid filled with trigger buttons. The large grid simultaneously displays the 32 trigger buttons for all 25 drum sounds.

Full view provides an overview of the whole pattern, as opposed to just one sequence. Because the trigger events for all sounds are shown, you can easily spot erroneous notes, and the entire pattern creation process is simpler and faster.



The selected sound is highlighted with a gray box in the step sequencer area, making it easy to set the velocity and gate time for each step, or offsets in Step mode (see Automating Parameter Values in Ultrabeat's Step Sequencer), within the context of all sequences/sounds in the pattern.

Both the grid and the Trigger and Gate/Length rows are displayed for the selected drum sound. This makes it simple to create trigger events in the full view grid, then set accents in the Gate/Length rows, for example.

Automating Parameter Values in Ultrabeat's Step Sequencer

The Edit Mode switch engages Ultrabeat's step automation feature. Step automation enables you to program parameter changes on a per-step basis—for each drum sound. You can adjust all parameters that can be automated, if desired, for each step.

Sound parameters that can be automated include all functions in the Synthesizer section except the menus (modulation routings and so on), buttons (oscillator type buttons, the buttons in the trigger/group section), and Pan/Spread parameters.

When you engage Step mode, Ultrabeat's interface changes in the following ways:

- Yellow frames appear around all parameters that can be automated in the Synthesizer section. Parameters that cannot be automated are still visible but are disabled.
- The velocity/gate row in the step grid changes to show the (parameter) offset row.

Tip: When creating offsets in Step mode, you may decide that you want to make a quick change to the original drum sound. Rather than switching back and forth between edit modes for a simple change, you can press Command-Option to temporarily flip Ultrabeat back into Voice mode.

All parameters that you choose to automate will appear in the pop-up menu at the top of the (parameter) offset row. Click the grid header (left of the zero axis) to open the menu.

Note: The simple act of moving a control element in the Synthesizer section will add the parameter to the offset menu, so take care.

Using Ultrabeat's (Parameter) Offset Row

This row enables you to view and enter offset values—on a per-step basis—for any of the Synthesizer section parameters that can be automated.

Parameter edits can be made in three ways:

- By directly adjusting the controls in the Synthesizer section
- By editing offset values in the offset row
- With the shortcut menu (see Using Ultrabeat's Parameter Offset Shortcut Menu Commands)

Note: Adjustments made in the offset row are relative to the current parameter value. Values shown in the offset row will be either added to, or subtracted from, the parameter value set in the Synthesizer section. In other words, parameter offsets increase or decrease—but do not specify—an absolute value for the parameter.

To create a parameter offset

1. Select the desired sound.
2. Click a position in the (parameter) offset row that corresponds to the step you want to edit.
3. Make the desired parameter changes in the Synthesizer section. Your changes will be recorded as an offset value for this step.
4. Repeat step 3 for each parameter that you want to edit for this step.

A parameter offset that has been created for a given parameter on a given step is represented in two ways.

- A yellow bar is drawn on the parameter that indicates the deviance (the offset) between the original parameter value and the new parameter value.
- In the (parameter) offset row, the offset from the original parameter is represented as a bar starting from the 0 point (horizontal center line).
  - Positive offsets are shown as a bar above the center line.
  - Negative offsets are shown as a bar below the center line.

Using Ultrabeat's Parameter Offset Shortcut Menu Commands

Control-click (or right-click) any step in the (parameter) offset row to open a shortcut menu. Choose one of the following commands:

- *Alter:* Changes the (selected) parameter values—for all steps—by a random amount.
- *Randomize:* Creates a new, random value for the selected parameter.

  Note: Consider saving your sequence/pattern before using either of the above commands.
- *Delete:* Deletes all steps for the currently selected parameter.

Muting, Soloing, and Resetting Ultrabeat Parameter Offsets

The (parameter) offset row offers the M(ute), S(olo), and Reset buttons.

- *M(ute):* Mutes the offsets of the selected parameter. This does not remove or reset existing offsets.
- *S(olo):* Enables you to hear the effect of your offsets on the selected parameter in isolation.

- *Reset:* All offset values for the selected parameter are set to 0 (no offset).
  - A second click on the Reset button removes the parameter from the offset menu.

  Note: The Reset button at the left of the velocity/gate row changes to Delete when clicked once. This Delete button mirrors the behavior of the Delete command: It deletes all steps for the currently selected parameter.

Exporting Ultrabeat Patterns as MIDI Regions

Patterns programmed in Ultrabeat's internal step sequencer can be exported as MIDI regions into the Arrange area of Logic Express.

To export an Ultrabeat pattern into the Logic Express Arrange area

1. Choose the desired pattern in Ultrabeat's Pattern menu.
2. Select the area to the left of the pattern menu.

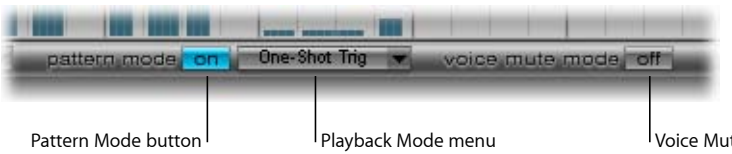
3. Drag the pattern to the desired position on the respective Ultrabeat track.

   A region is created, containing all MIDI events, including Swing and Accent parameter settings. Accents are interpreted as polyphonic pressure events. Step automation events, created in Step mode, are also exported as part of the MIDI region.

   Note: To avoid double-triggering while playing back the exported MIDI region, you may want to turn off Ultrabeat's internal sequencer.

Using MIDI to Control Ultrabeat's Sequencer

Pattern performance can be influenced by incoming MIDI notes. This allows you to spontaneously interact with the step sequencer, making Ultrabeat an excellent live performance instrument. The way Ultrabeat reacts to MIDI control is determined by the chosen pattern, playback, and voice mute mode options.



- *Pattern Mode (On/Off) button:* When enabled, allows you to choose and start patterns with MIDI note messages. The Transport button turns blue to indicate that Ultrabeat is ready to receive incoming control commands.
  - MIDI notes C-1 to B0 switch between patterns: C-1 selects pattern 1, C#-1 selects pattern 2, and so on up to pattern 24, selected when MIDI note B0 is received.
- *Playback Mode menu:* Determines pattern playback behavior when an incoming MIDI note is received. You can choose one of the following options:
  - *One-Shot Trig(ger):* The reception of a MIDI note starts the pattern, which plays once through its cycle, then stops. If the next note is received before the pattern has reached its final step, the new note stops playback of the first pattern and the next pattern begins playing immediately—this can be a different pattern or the same pattern, depending on the MIDI note received. Note-off events are ignored.
  - *Sustain:* The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the corresponding MIDI note is released (a note-off event is received).
  - *Toggle:* The reception of a MIDI note starts the pattern and it continues playing in an infinite loop until the next note is received. If it is the same note, the pattern stops immediately. If it is a different note, the sequencer immediately switches to the new pattern.

  Note: Toggle mode allows you to switch between patterns in the middle of a bar—the sequencer stays in time and automatically jumps to the corresponding beat of the new pattern. This isn't the case in One-Shot Trig mode, which starts the new pattern from the beginning as soon as you play a MIDI note.
- *Toggle on Step 1:* The behavior is the same as Toggle mode except that the pattern change or stop occurs the next time beat 1 is reached—at the beginning of the next pattern cycle.

- *Voice Mute Mode button:* When active, playing MIDI note C1 and above mutes the corresponding sound in the Ultrabeat mixer. A subsequent MIDI note of the same pitch unmutes it. This is ideal for spontaneous rearranging of patterns and/or muting single elements of a pattern without deleting them. This is especially useful in a live performance or remixing situation.

All the creative pattern switching options discussed in this section are achieved through the use of MIDI note messages and can therefore be simply recorded, edited, arranged, and automated in Logic Express.

Ultrabeat Tutorial: Introduction

The Ultrabeat tutorials cover a number of specific sound creation tips. These tutorials will help you explore the vast and complex possibilities available to you in Ultrabeat. You'll discover that there is hardly a category of electronic drum sound that Ultrabeat can't create easily.

Note: In Ultrabeat's Settings > 03 Tutorial Settings folder, you will find Tutorial Kit, a drum kit that contains all the drum sounds discussed in the tutorials. The Tutorial Kit also includes the Standard Tut(orial) drum sound, which is a default set of neutral parameters that provide an excellent starting point for many of the examples.

All Ultrabeat tutorial sections are listed below:

- Ultrabeat Tutorial: Creating Kick Drums
- Ultrabeat Tutorial: Creating Snare Drums
- Ultrabeat Tutorial: Creating Toms and Tonal Percussion
- Ultrabeat Tutorial: Creating Hi-Hats and Cymbals
- Ultrabeat Tutorial: Creating Metallic Sounds
- Ultrabeat Tutorial: Creating Extreme Sounds
- Ultrabeat Tutorial: Programming in Building Blocks

Ultrabeat Tutorial: Creating Kick Drums

Electronically produced kick drum sounds are based primarily on the sound of a deeply tuned sine wave.

To program a kick drum in Ultrabeat

1 Choose Settings > 03 Tutorial Settings > Tutorial Kit, and select Standard Tut from the Assignment section.

Note that Oscillator 1 is in Phase Oscillator mode.

2 Find a suitably tuned pitch in the lower octaves by soloing the bass drum along with other important tonal elements of the song (a bass or pad sound, for example). Drag the Osc 1 Pitch slider to adjust the pitch until appropriate.

3 Use Env 4 to shape the volume of the bass drum.

For slower beats you'll want a longer decay phase, whereas at faster tempos you'll choose a shorter decay time. The attack time of Env 4 should be very short in any case (0, in most cases) or the sound will lose its percussiveness and its ability to be clearly heard in the mix.

The kick drum still sounds very soft and is somewhat reminiscent of the famous TR-808 bass drum. It's still missing a clearly defined attack.

To give the bass drum more kick by controlling the pitch with an envelope

1 Ensure that Env 1 is chosen in the Mod menu of Oscillator 1's Pitch parameter.

2 Set the degree of modulation by dragging the blue Mod slider approximately 3 to 4 octaves above the original pitch.

3 Set the attack time in Env 1 to 0 by dragging the leftmost of the two junction points on the x-axis all the way to the left.

4 Experiment with the decay time by dragging the rightmost of the two junction points on the x-axis; you'll discover that higher decay values (shifting the Bezier handle to the right) result in sounds similar to synth toms, whereas shorter decay values (shifting to the left) provide the kick character.

5 Change the Mod amount (the blue slider) of Osc 1 Pitch again (see step 2).

The interaction of this parameter with the envelope's decay time provides numerous possibilities for shaping the kick or punch of the bass drum sound.

Note: This simple bass drum sound is listed as Kick 1 in the Tutorial Kit, at a pitch of C1.

Ultrabeat Tutorial: Reducing Tonality

One advantage of bass drums based on sine waves is that their sound can be precisely tuned to match the song. The disadvantage is that a recognizable pitch is not always desirable. Ultrabeat offers several methods to reduce the tonality of the sound. A very effective tool is the 2 Band EQ.

To reduce tonality using the 2 Band EQ

1. For band 1, select the Shelving mode at a frequency of about 80 Hz, a high Q value, and a negative Gain value.
2. For band 2, select the Peak mode at a frequency of around 180 Hz, a medium Q value, and also a negative Gain value.

   On the EQ graph, notice how the frequencies around 80 Hz are boosted, while the surrounding frequencies are reduced.

3. Vary the frequency of band 2 (easily recognizable in the blue part of the EQ graph) to influence the extent of bass drum tonality.

   A further method for reducing the tonality of a drum sound that is rich with overtones is to use a lowpass filter. In the following example you will control the cutoff frequency of the filter with an envelope.

To reduce tonality using a lowpass filter

1. Reload the Standard Tutorial sound, choose A#0 as the basic pitch in Oscillator 1, and modulate it with Env 1.
2. Increase the Saturation parameter value to enhance the overtones of the drum sound.

   Note that the output of Osc 1 is directed to the filter, as the filter bypass button (the arrow between Osc 1 and the filter) is activated.
3. Set Filter type to LP 24.
4. Set Cutoff value to 0.10.
5. Set Mod Source for Cut to Env 3.

6 Set Mod Amount for Cut to 0.60.

7 Set Resonance to 0.30.

8 Set the attack time of Env 3 to 0. Use the Decay time of Env 3 to shape the sound of the filtered bass drum.

9 You may also choose to control the filter resonance with an envelope. Make sure you dedicate a single envelope to this function (in this case, use Env 2 as a Mod source for Res). Choose a Mod amount for Res of about 0.80. Select a longer decay time in Env 2 than in Env 3 and listen carefully to the fatter and more atonal bass drum sound achieved through this Res modulation (due to the higher filter resonance).

Note: The bass drum described in the above example is listed as Kick 2 in the Tutorial Kit, at a pitch of C#1. It also features an interesting EQ setting, as described in the next section.

Ultrabeat Tutorial: Adding Bass, Kick, and Body to Sounds

Try the options outlined in this section to add "missing" elements to your sounds.

Adding some bass to your sound:
Use the Kick 2 filtered bass drum sound as a starting point, and try out the remaining parameters in the phase oscillator. You will discover that high saturation values make the sound rounder and add more bass, for example. The character of the example is beginning to head in the direction of a TR-909.

Enhancing the attack transients of your sound:
To get even closer to the TR-909, use an EQ setting as shown in the following figure. Note that the low frequency pressure point around 60 Hz (in the red area on the EQ graph) as well as the assertive punch or kick (the blue area starting at 460 Hz and up) of a 909 bass drum are strengthened. (This EQ setting is already part of the Kick 2 setting.)

Using envelopes to change the color of your sound:
In the example, all four envelopes are being used. Take some time to play with the shapes of the envelopes, while maintaining the attack and decay settings. Experiment with the junction points of the decay phase in the different envelopes to familiarize yourself with the sound-shaping options available. Start with the decay phase of Env 4, which controls both the volume of Oscillator 1 and filter resonance, and observe how reshaping the belly of the envelope can change the character of the sound from crisp and short to round and full.

Ultrabeat Tutorial: Creating the Ultrabeat Kick

You can create bass drum sounds that are uniquely "Ultrabeat." Try modulating pitch with an LFO instead of an envelope, for example.

To create an LFO-modulated kick drum

1. Start with the Standard Tutorial sound at a pitch of A#0 (Osc 1 Pitch), and choose LFO 1 as the Mod source in the Osc 1 Pitch section.
2. Set the degree of modulation by dragging the blue Mod slider to a value of A3.
3. Set LFO 1 to a low number of Cycles (25 to 35), a high Rate (starting with 70 Hz and higher) and a medium value for Decay (set the Ramp knob to about –190).
4. Experiment with the LFO waveform and you'll discover that you can attain different nuances in the character of the bass drum attack.
5. Modulate the Asym(metry) parameter with the same LFO, and also vary the Slope and Saturation values.

This method enables you to create very different bass drum sounds with a single oscillator, one LFO, and one envelope (for volume). The character of the sounds can range from soft to punchy, and the degree of tonality in the sound can be adjusted to taste.

Note: The bass drum sound described is listed as Kick 3 in the Tutorial Kit, at a pitch of D1.

Use the second oscillator (with similar settings or with a sample), or use the filter and the ring modulator—the sky's the limit as far as your imagination is concerned, so get on with it and create that next "gotta have it" drum sound.

Note: You can find an "emulation" of the legendary 808 bass drum in Kick 4 in the Tutorial Kit, at a pitch of D#1.

Ultrabeat Tutorial: Creating Snare Drums

The sound of an acoustic snare drum consists primarily of two sound components: the sound of the drum itself and the buzzing of the snare springs. Try to approximate this combination in Ultrabeat with a single oscillator and the noise generator.

To create a basic snare drum

1. Load the Standard Tutorial setting. Deactivate Oscillator 1, and switch Oscillator 2 on (in phase oscillator mode).
2. Choose LFO 1 in the Mod menu of Osc 2 Pitch.
3. Set the pitch value for Osc 2 to around G#2 and set the Mod amount (the blue Mod control) to about 3 to 4 octaves higher.

   You have modulated Osc 2 Pitch with a rapidly vibrating LFO with a medium Ramp Decay value. This eliminates the sine wave—which is not especially desirable for a snare sound, in contrast to the bass drum.
4. Set LFO 1 to a high Rate. Choose a value of 20 for Cycles and –20 for Ramp. Set the LFO Waveform parameter to a value of about 0.58, which is a square wave.
5. Use Env 1 to control the volume of Oscillator 2 by setting Vol to the lowest possible value (–60 dB), choosing Env 1 in the Mod pop-up menu, and adjusting the modulation intensity to a point just below its maximum value.

The figure shows the settings of Oscillator 2 and Env 1.

6 Experiment with different Slope and Asym values to impart a more or less electronic character to the sound.

7 Turn on the noise generator and control its volume with the same quick envelope used in Osc 2 Volume.

8 Use the filter parameters of the noise generator to roughen up, refine, or add bright frequencies to the noise component of the snare drum sound. Select an LP filter type, and try a filter frequency between 0.60 and 0.90. Modulate it with LFO 1, which you're already using to control the pitch of Oscillator 2.

Note: The snare drum sound is listed as "snare 1" in the Tutorial Kit, at a pitch of E1.

To refine the snare drum sound using FM synthesis

1 Turn on Oscillator 1 in FM mode. Use Env 1 to control the volume of Osc 1 as well.

2 Choose a pitch for Oscillator 1 that's about an octave lower than Oscillator 2. Consciously avoid even intervals between the oscillators and detune them slightly from each other. For example, try a pitch setting of F#2 in Osc 2 and E1 in Osc 1, then fine-tune Osc 1 a few cents higher by holding Shift while adjusting the Osc 1 Pitch slider.

3 Experiment with FM Amount, and add more tone (low FM Amount) or noise (more FM Amount) as desired. Also try modulating the FM Amount with a fast LFO.

Note: An exemplary snare drum sound that uses FM can be found in the Tutorial Kit at a pitch of F1. It is listed as "snare 2."

Higher FM Amount values lead to considerably more overtones and a very electronic sound character. If you want to make the sound more acoustic, feed Oscillator 1 (and possibly Oscillator 2 as well) into the main filter. Use these settings to start: LP 24 mode, and a Cutoff value of about 0.60.

Ultrabeat Tutorial: Recreating the TR-808 Snare

The famous 808 snare is based on two resonating filters and a noise generator, fed through a highpass filter. The mix ratio of the two filters and the volume of the noise generator can be adjusted. This structure cannot be 100% replicated in Ultrabeat.

To clone the 808 snare sound

1. Load the Standard Tutorial setting.

   You are now ready to replicate the resonating filters of the 808 snare using two cleverly programmed phase oscillators.

2. Assign slightly different Slope values to two phase oscillators, and detune them by almost an octave.

3. Adjust the tonal relationship between the oscillators so that it is uneven—from E3 to F2, for example.

4. Control the volume of each oscillator with a different envelope. Adjust the decay times so that the envelope for the lower-tuned oscillator has a longer decay time than the very snappy envelope setting for the higher oscillator.

5. Feed the output of both oscillators into Ultrabeat's main filter, and hollow out the sound with a highpass filter. Activate the filter bypass button in both oscillators. Choose the HP 12 setting in the filter, a Cutoff value around 0.40, and a Resonance value of about 0.70.

   You have just cleverly emulated both of the 808's resonating filters. Shifting the pitch of both oscillators simulates the behavior of the 808's Tone control, by the way.

To complete the 808 emulation by adding some noise

1. Switch the noise generator on, and activate the highpass mode in its filter (HP).

2. Set the Cutoff value to about 0.65, Resonance to 0.35, and add a little Dirt (around 0.06).

   The noise generator provides the sustained snare sound. It should be shaped by its own envelope—independent of the decay phases of both oscillators—to get 808-like results. Changing the volume of the noise generator simulates the snap parameter of the 808.

   Note: The 808 snare drum described is listed as "snare 3-808" in the Tutorial Kit, at a pitch of F#1. It also features an interesting EQ setting.

Ultrabeat Tutorial: Changing Dynamics with Velocity

Use the 808 snare drum sounds in the Tutorial Kit to explore the possibilities Ultrabeat offers for implementing velocity.

To use velocity modulation

1. Select the "snare 3-808" sound.

2 Open the “via” pop-up menu below the Oscillator 1 Volume knob and choose Vel. A slider appears on the ring around the knob.

3 Drag the slider clockwise. When you drag the slider, a help tag displays the value. Set it to 0 dB.

4 Repeat steps 2 and 3 in both Oscillator 2 and the noise generator.

You can now dynamically play the sound using velocity.

To increase the performance dynamics

1 Reduce the values of the individual volumes by turning down the Volume knobs in both oscillators and the noise generator. Note how the Mod ring and its Via sliders also move back. Change the Via slider positions until all three Volume knobs look like this:

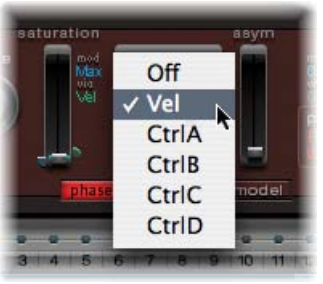
If you use differing intensities for each Volume knob when completing this step, you'll have the potential of individual velocity reactions for each sound component.

2 Increase the dynamics of the sound as a whole by assigning the following setting to the Voice Volume knob:

You now have an 808 snare that is exceptionally responsive to velocity. As you may know, this wasn't possible with the original—not even an 808 sample could offer the dynamic volume control of individual sound components that's demonstrated here. A sample offers you only the whole sound, not its constituent parts.

In the next step, you use velocity to control the character of the sound—individually for each component—plus volume, of course.

3 In the Saturation Mod pop-up menu of Oscillator 2, choose Max, and then choose Vel(ocity) in the corresponding Via pop-up menu.



4 Set the additional control that appears as shown in the figure below, to control the character of the sound with velocity:

5 Repeat this with the other parameters of Oscillator 2, as well as pitch:

6 Modulate the noise generator as follows:

- *Cut parameter:* Choose Max as modulation source, then set the modulation control as shown below.
- *Dirt parameter:* Choose LFO 2 as modulation source, then set the modulation control as shown below.

The sound is now nothing like an 808 snare, which was your goal. Keep experimenting with velocity and figure out when it makes sense to use it as a direct or indirect modulation source, in either its positive or negative form.

Note: The dynamic 808 snare drum described above is listed as “snare 4-vel” in the Tutorial Kit, at a pitch of G1.

Ultrabeat Tutorial: Recreating the Kraftwerk Snare

Another classic electronic snare drum sound is the highly resonant lowpass filter of an analog synthesizer that quickly closes with a snap. This sound was used extensively by Kraftwerk.

To recreate the Kraftwerk snare sound with Ultrabeat

1. Select the Snare 1 sound.
2. Direct the signals of both oscillators and the noise generator to the main filter.
3. Modulate Cutoff with Env 1 (which is already modulating the volume of the noise generator).
4. Modulate the filter resonance with Env 2.
5. Experiment with the parameters mentioned in steps 2 to 4 (especially the envelopes), introduce EQ into the sound, and discover how much "playing room" these basic settings allow you.

Note: An exemplary sound is listed as "snare 5-KW" in the Tutorial Kit, at a pitch of G#1. Analyze this sound, and compare it to your own creation.

Ultrabeat Tutorial: Creating Toms and Tonal Percussion

Tonal percussion sounds such as toms or congas are relatively easy to emulate electronically with sine or triangular wave oscillators. Ultrabeat's phase oscillator offers you a broad spectrum of suitable basic sounds with which to start. Control the pitch of the oscillators with envelopes, and use the programming techniques discussed in the bass and snare drum sections to alter tonality. You should find it easy to create a broad range of toms and similar sounds.

Note: At the pitches A1 to B0 in the Tutorial Kit you'll find typical 808 toms. Analyze these sounds and modify them as you see fit.

Spend some time experimenting with the Model mode of Oscillator 2. Try to familiarize yourself with the effect of each parameter, and create some of your own tonal percussion sounds, ranging from small tabla drums to glass bowls.

Note: The "tabla" and "glass" sounds (at pitches C2 and C#2) in the Tutorial Kit combine both Osc 2 Model and FM. They are also good examples of the complex use of velocity as a modulation source.

Ultrabeat Tutorial: Creating Hi-Hats and Cymbals

Electronic hi-hat sounds are easy to create in Ultrabeat.

To create a hi-hat in Ultrabeat

1. Load the Standard Tutorial sound.

2 Switch off Oscillator 1 and turn on the noise generator.

3 In the noise generator, make sure the Cutoff parameter is modulated by Env 1, the modulation is negative, and the position of the Mod slider is below that of the base parameter value.

4 Use rather short decay values for Env 1 and Env 4.

5 Set the attack time of Env 4 to a value of 0. The attack time of Env 1 should also be rather short, but not equal to 0.

Note: You'll find a similarly constructed sound listed as "hihat 1" at a pitch of F2 in the Tutorial Kit. Also analyze the hi-hat sound "hihat 2" at pitch F#2.

It's not far from the hi-hat to the crash cymbal. The main difference between a hi-hat and crash cymbal sound is the length of the decay time. Correct assignment of the envelopes is the key to producing different cymbal sounds.

Select the Cym 1 and Cym 2 sounds in the Tutorial Kit and try different envelope assignments and settings for Cutoff and Volume in the noise generator, Cutoff and Volume in the main filter, and so on.

Ultrabeat Tutorial: Creating Metallic Sounds

If you want to create metallic sounds with Ultrabeat, the ring modulator and the Model oscillator are the ideal tools.

To use the ring modulator

1 Load the Standard Tutorial sound.

2 Activate a phase oscillator and the Model oscillator. Choose a pitch for each oscillator above C3 so that a slightly detuned interval is created.

3 In the Material Pad of the Model oscillator, choose a setting with plenty of overtones, as in the figure below.

4 Set the volume of each oscillator to a value of –60 dB, and click "ring mod" to turn on the ring modulator.

You've just created a bell-like sound that you can filter with a high resonance value if required.

Note: You can find a similar sound listed as Ring Bell at a pitch of A2 in the Tutorial Kit.

Ultrabeat Tutorial: Creating Extreme Sounds

Ultrabeat features extremely fast envelopes and uncommonly powerful LFOs. Use these modulation sources to perform extreme modulations of the oscillator and filter parameters.

To create "out of the ordinary" sounds

- Try modulating as many targets as possible.
- Don't be afraid of using extreme settings.
- Use a quick envelope to drive the filter to self oscillation for a fraction of a second.
- Use a few LFO cycles at a much higher rate than other cycles.
- Experiment with the Dirt parameter or the bit crusher.

Ultrabeat Tutorial: Programming in Building Blocks

As you become familiar with drum sound programming, you may begin thinking in building blocks, realizing that drum sounds usually consist of different components.

After you mentally—or physically—write down your list of components, try to emulate each component that contributes to the sound's character, making use of the different sound generators available in Ultrabeat. Assigning dedicated amplitude envelopes to the different components allows you to control their temporal behavior individually. For example, you can emulate the body of a drum with Oscillator 1, the sound of the stick hitting the skin (or first transient) with the noise generator, and additional overtones and harmonics can be provided by Oscillator 2 or the ring modulator.

When you begin thinking that drum sounds consist of several building blocks or layers, the design of the Volume controls in the individual sound generators might make more sense to you—this is the place where the blocks are combined, balanced, and controlled.

GarageBand Instruments

12

GarageBand Instruments are automatically installed with Logic Express. You use GarageBand instruments as you would other software instruments in Logic Express.

This chapter covers the following:

- GarageBand Instrument Features (p. 322)
- GarageBand Analog Basic (p. 323)
- GarageBand Analog Mono (p. 324)
- GarageBand Analog Pad (p. 325)
- GarageBand Analog Swirl (p. 326)
- GarageBand Analog Sync (p. 328)
- GarageBand Bass (p. 329)
- GarageBand Church Organ (p. 330)
- GarageBand Digital Basic (p. 331)
- GarageBand Digital Mono (p. 332)
- GarageBand Digital Stepper (p. 333)
- GarageBand Drum Kits (p. 334)
- GarageBand Electric Clav(inet) (p. 335)
- GarageBand Electric Piano (p. 335)
- GarageBand Guitar (p. 336)
- GarageBand Horns (p. 337)
- GarageBand Hybrid Basic (p. 338)
- GarageBand Hybrid Morph (p. 340)
- GarageBand Piano (p. 342)
- GarageBand Sound Effects (p. 343)
- GarageBand Strings (p. 344)
- GarageBand Tonewheel Organ (p. 345)

- GarageBand Tuned Percussion (p. 346)
- GarageBand Voice (p. 347)
- GarageBand Woodwind (p. 348)

GarageBand Instrument Features

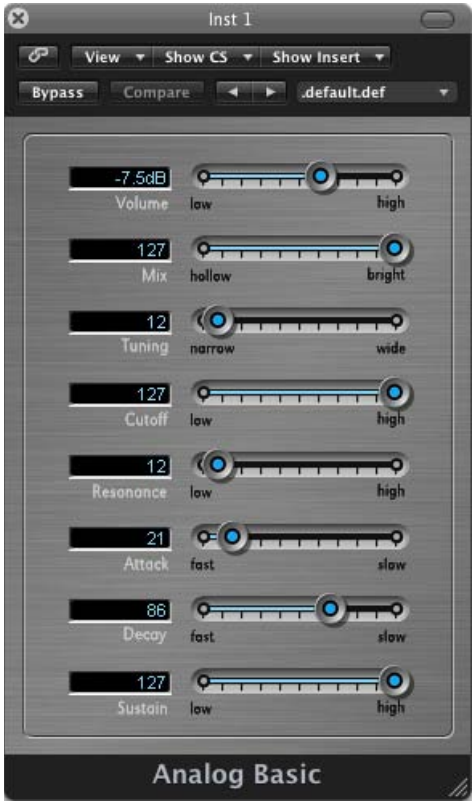
GarageBand Instruments are actually less CPU and memory-intensive versions of equivalent Logic Express instrument plug-ins. The GarageBand synthesizers use a cut-down version of the ES2. Similarly, organ sounds are generated by a feature-limited EVB3, and electric piano sounds are emulated by an instrument based on Logic Pro's EVP88. Other sounds, such as strings, brass, drum kits, and so on, make use of a simplified sample player that is based on the EXS24 mkII.

The interface of all GarageBand instruments is limited to a few key parameters, simplifying their use. The available parameters were carefully chosen to provide maximum impact and flexibility, making it easy and fast to create great sounds.

The inclusion of these instruments makes it simple to import GarageBand project files into Logic Express. After you import a GarageBand project, you will have access to the advanced sequencing, mixing, and effect processing options in Logic Express. You can also choose to replace the instruments in your GarageBand project with any of the Logic Express instruments.

GarageBand Analog Basic

The Analog Basic sound is based on the ES2. This is a simple analog synthesizer tone that is useful for a range of musical styles.

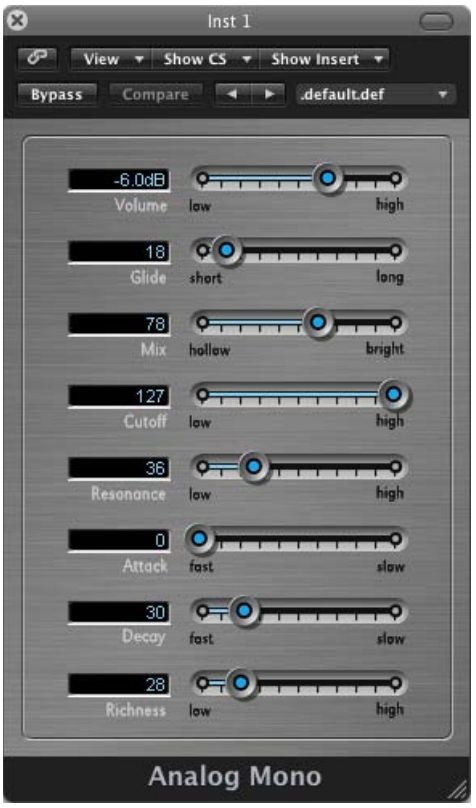


- *Volume slider:* Sets the overall volume level of the instrument.
- *Mix slider:* Determines the mix level of (balance between) the oscillator signals.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.

- *Sustain slider:* Determines the level of the sound after the Attack and/or the Decay phase has completed.

GarageBand Analog Mono

This is a monophonic (one note can be played at a time) analog synthesizer lead sound that is based on the ES2.

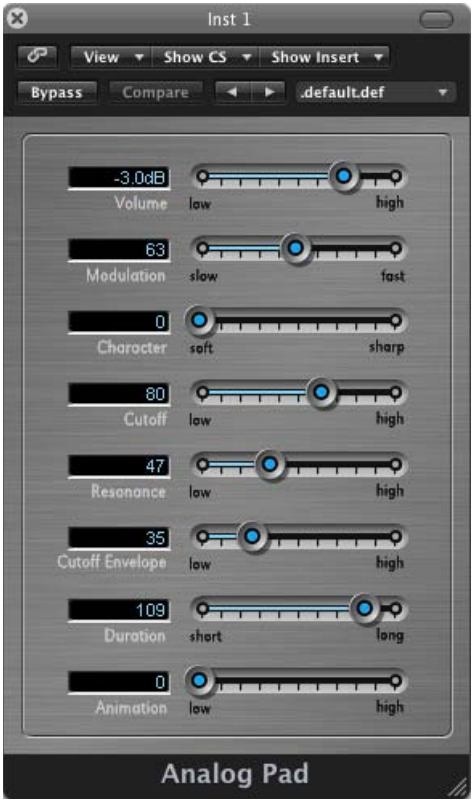


- *Volume slider:* Sets the overall volume level of the instrument.
- *Glide slider:* Determines the time it takes a note pitch to change (slide) to another note pitch.
- *Mix slider:* Determines the mix level of (balance between) the oscillator signals.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.

- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Richness slider:* Determines the complexity of the sound texture, making the sound fuller.

GarageBand Analog Pad

The Analog Pad is based on the ES2. This is a warm analog synthesizer pad that is useful for a range of musical styles.

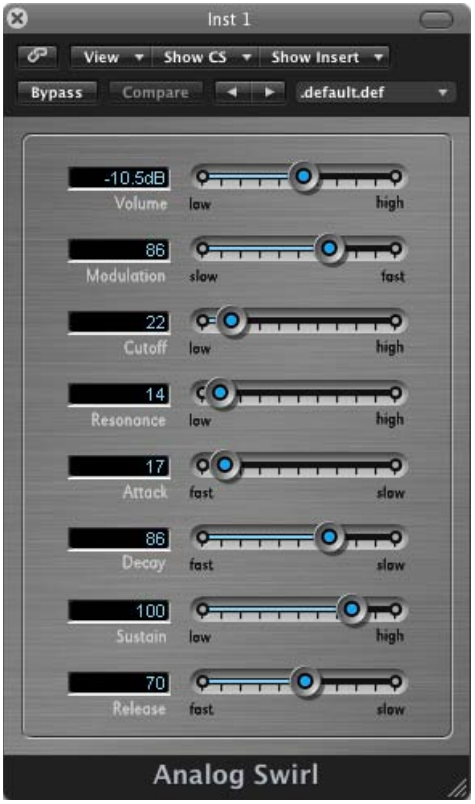


- *Volume slider:* Sets the overall volume level of the instrument.
- *Modulation slider:* Makes the sweeping movement of the pad faster or slower.
- *Character slider:* Determines if the sound is soft or sharp.

- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determines the strength of the sweeping movement.
- *Duration slider:* Determines the duration of the sweeping movement.
- *Animation slider:* Determines the effect on the pad sound by the envelope.

GarageBand Analog Swirl

The Analog Swirl instrument is based on the ES2. It is a warm analog synthesizer pad that features a chorused, swirling sound.

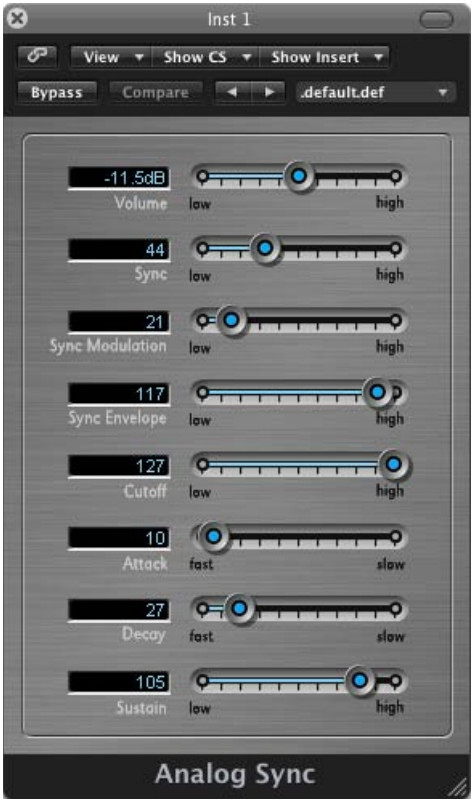


- *Volume slider:* Sets the overall volume level of the instrument.
- *Modulation slider:* Makes the sweeping movement of the pad faster or slower.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.

- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack and/or the Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Analog Sync

The Analog Sync instrument is based on the ES2. It emulates tones from analog synthesizers that synchronize two oscillators to produce their sound. The Analog Sync instrument is most useful for hard-edged analog synthesizer lead sounds.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Sync slider:* Determines the synchronization (or lack of it) between the two oscillators, and therefore the harshness of the sound.
- *Sync Modulation slider:* Determines how much the synchronization of the two oscillators is modulated, resulting in more complex (and harder) tones.
- *Sync Envelope slider:* Determines the amount that envelope parameters affect the sound.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.

- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack and/or the Decay phase has completed.

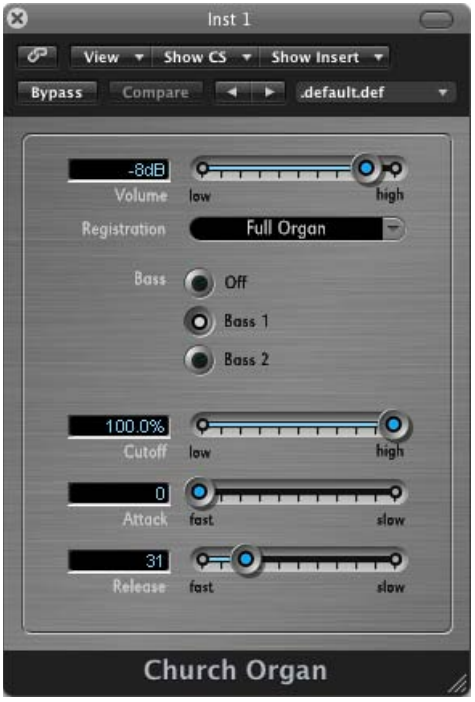
GarageBand Bass

The Bass instrument is sample based. It emulates electric and acoustic basses.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Church Organ

The Church Organ instrument is sample based. It emulates a pipe organ.

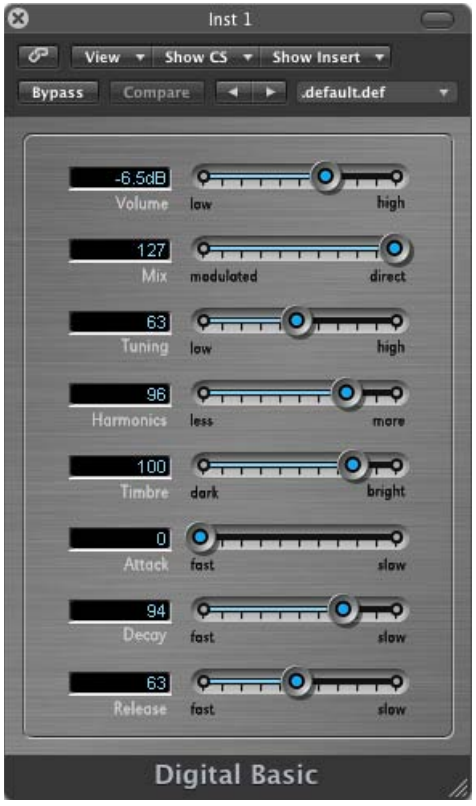


- *Volume slider:* Sets the overall volume level of the instrument.
- *Registration pop-up menu:* Provides a number of preset registrations. Registrations are combinations of different pipe organ stop (lever) settings that change the tonal character of the sound by enabling or disabling particular pipes. This alters the harmonics that are heard when you play a key.
- *Bass buttons:* You can activate the lower (bass) pipes with these buttons, adding these lower harmonics, which makes your sound richer and fuller.

  Note: The bass buttons are not available in all registrations.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Digital Basic

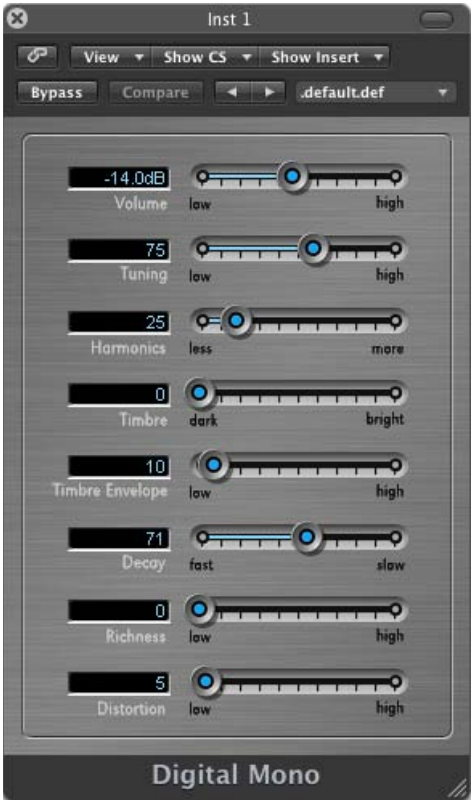
The Digital Basic instrument is based on the ES2. This is a simple digital synthesizer sound that is useful for a range of musical styles.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Mix slider:* Mixes two tones together.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Harmonics slider:* Increases or decreases the number of harmonics (overtones) in the sound. This can change the sound dramatically or subtly, so feel free to experiment.
- *Timbre slider:* Changes the color of the sound from dark to bright.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Digital Mono

The Digital Mono instrument is based on the ES2. This is a monophonic digital synthesizer lead sound.



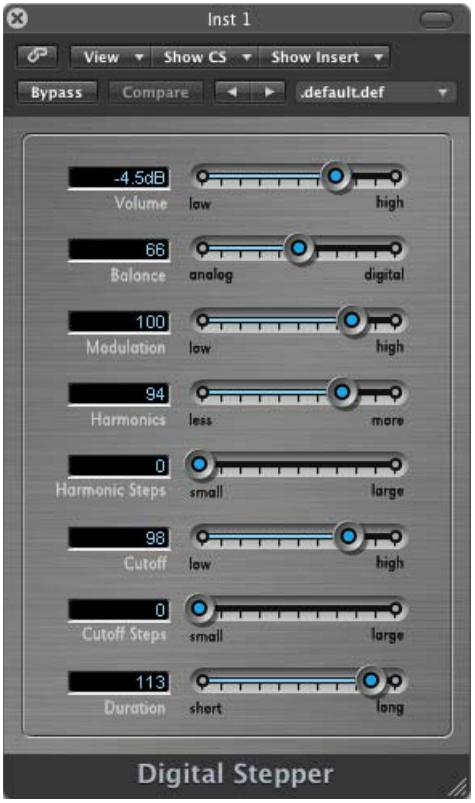
- *Volume slider:* Sets the overall volume level of the instrument.
- *Tuning slider:* Sets the overall pitch of the instrument.
- *Harmonics slider:* Makes the sound a little thicker (more) or thinner (less).
- *Timbre slider:* Changes the color of the sound from dark to bright.
- *Timbre Envelope slider:* Dynamically changes the color of the sound, depending on how firmly you strike your keyboard.
  - Low values have little or no effect on the color of the sound, no matter how hard you play the keys.
  - High values result in significant changes in the sound, in response to firmer or softer keyboard playing.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.

- *Richness slider:* Subtly detunes each played note from one another, making the sound a little thicker, particularly when *high* parameter values are used.
- *Distortion slider:* Distorts the overall sound, making it quite nasty and aggressive.

  Important: Take care with the Distortion parameter as it can significantly increase the overall volume of the instrument, which may result in damage to your speakers or ears!

GarageBand Digital Stepper

The Digital Stepper instrument is based on the ES2. This is a digital synthesizer that can step through a number of tones, creating a rhythmic pattern.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Balance slider:* Sets the balance between a harder and more spiky sound (digital) and a warmer, softer sound (analog).
- *Modulation slider:* Applies more or less modulation, making the sound more lively when high settings are used.
- *Harmonics slider:* Makes the sound a little thicker (more) or thinner (less).

- *Harmonic Steps slider:* Makes the tonal steps more (large) or less (small) noticeable.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Cutoff Steps slider:* Sets the amount of cutoff applied to each step. A higher value (large) will make the cutoff effect more pronounced.
- *Duration slider:* Sets the length of the steps.

GarageBand Drum Kits

The Drum Kits are sample based. Kits emulated include rock, pop, jazz, electronic, orchestral, and Latin, among others.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Electric Clav(inet)

The Electric Clavinet sound is based on the EVD6. It emulates the Hohner D6 clavinet.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Damper slider:* Changes the tone of the clavinet, making it less sustained and more woody sounding as you move towards the *high* setting.

GarageBand Electric Piano

The Electric Piano sound is based on the EVP88. It sounds like the Fender Rhodes and Wurlitzer electric pianos.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Model buttons:* A more bell-like tone is achieved when the Tines button is selected.
- *Decay slider:* A short value makes the sound almost plucked, whereas a long setting makes the sound sustain while the keys are held.

GarageBand Guitar

The Guitar sound is sample based. It emulates a number of different acoustic and electric guitar sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

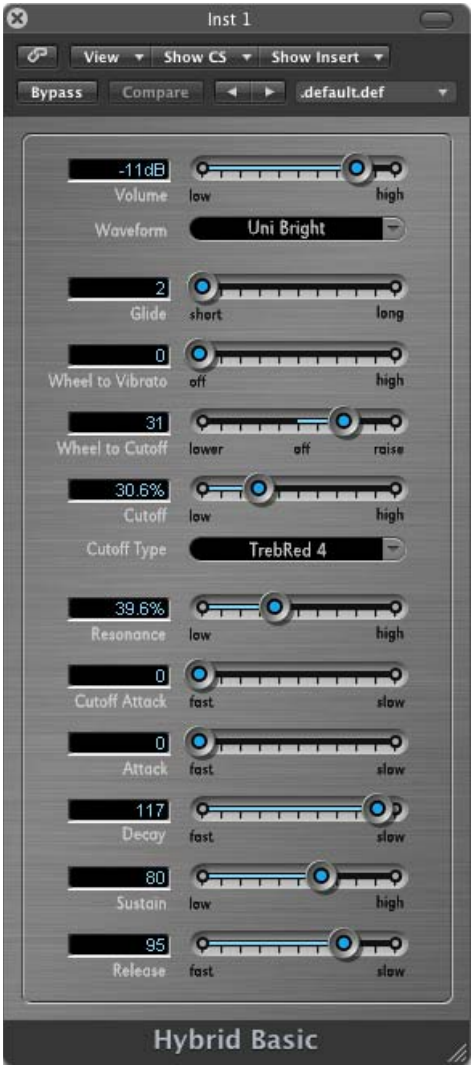
GarageBand Horns

The Horns instrument is sample based. It emulates several brass sections and a number of individual brass instruments.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Hybrid Basic

The Hybrid Basic instrument is a sample-based synthesizer that can create spectacular sounds.

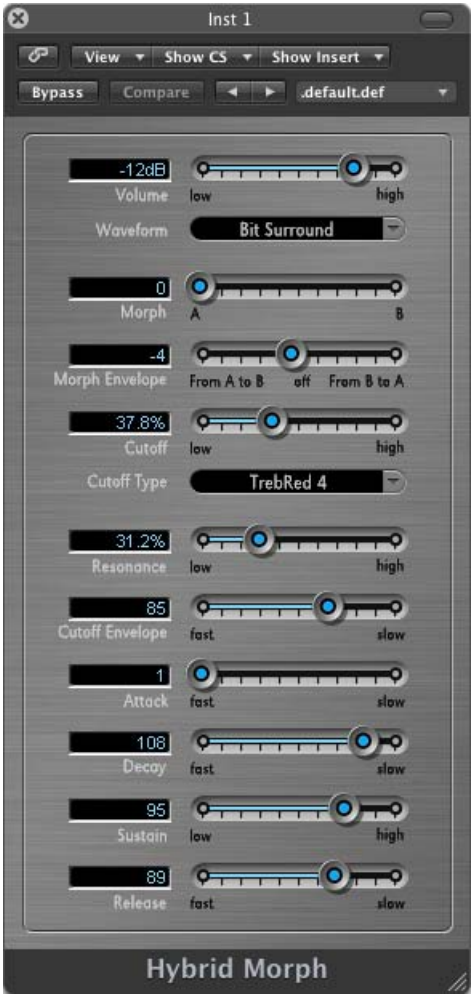


- *Volume slider:* Sets the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Glide slider:* Determines the time it takes a note pitch to change (slide) to another note pitch.

- *Wheel to Vibrato slider:* Determines the amount of pitch modulation by your keyboard's modulation wheel.
- *Wheel to Cutoff slider:* Determines the depth of Cutoff modulation by your keyboard's modulation wheel.
- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Cutoff Type pop-up menu:* Enables you to choose from a number of preset filter curves. Try them out, and experiment with the Cutoff and Resonance parameters.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Attack slider:* Determines the time it takes before the Cutoff parameter begins to affect the sound.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack and/or the Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Hybrid Morph

The Hybrid Morph instrument is a sample-based synthesizer that can create spectacular sounds. It differs from Hybrid Basic in that each waveform is based on two sample layers. This leads to a different sonic character.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Waveform pop-up menu:* Choose the sample set used to generate the basic synthesizer sound.
- *Morph slider:* Controls crossfades between the two sample layers.
- *Morph Envelope slider:* Controls the morph over time. For example, setting the Morph parameter to B and the Morph Envelope to "From A to B" morphs the Wave from A to B, in accordance with the ADSR envelope settings.

Note: If you set the Morph parameter to A and the Morph Envelope to "From A to B," certain ADSR settings will result in no sound. In this context, you can achieve interesting results by using the modulation wheel to offset the Morph parameter during live performances.

- *Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Cutoff Type pop-up menu:* Enables you to choose from a number of preset filter curves. Try them out, and experiment with the Cutoff and Resonance parameters.
- *Resonance slider:* Emphasizes the frequency range around the point determined by the Cutoff parameter.
- *Cutoff Envelope slider:* Determines the strength of the envelope shaping applied to the Cutoff parameter.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Decay slider:* Makes the harmonic portion of the sound (bright) sustain for a longer time at slow values. Faster values move to the Sustain level more quickly.
- *Sustain slider:* Determines the level of the sound after the Attack and/or the Decay phase has completed.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Piano

The Piano instrument is sample based. It emulates a number of classical and jazz piano sounds. It also emulates several accordions and a harpsichord, and it provides a number of pad sounds.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

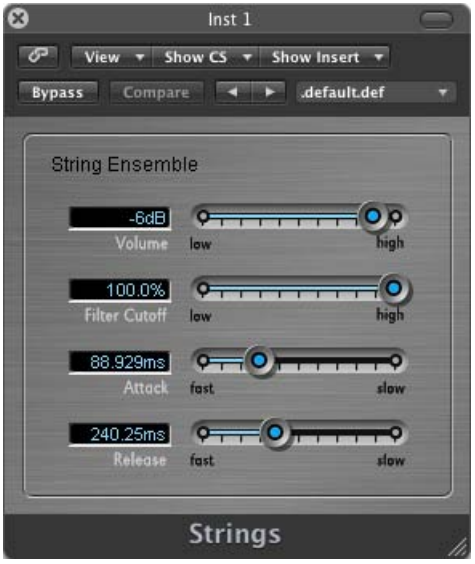
GarageBand Sound Effects

The Sound Effects are sample based. These cover a number of nature sounds, laughter and applause, and so on.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Strings

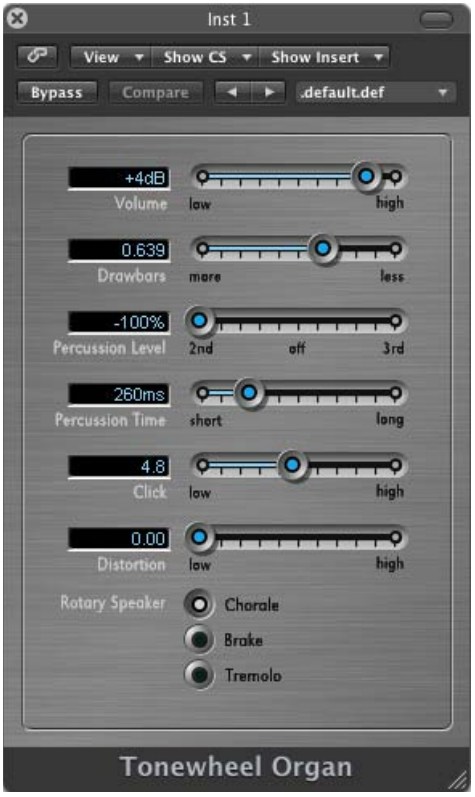
The Strings instrument is sample based. It emulates string sections and a number of individual stringed instruments. These include violins, violas, cellos, harps, and several traditional instruments, such as sitar, koto, and zither.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Tonewheel Organ

The Tonewheel Organ sound is based on the EVB3. It emulates the Hammond B3 organ, but it is also capable of emulating Farfisa, Wurlitzer, and other electric organs.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Drawbars slider:* Increases or decreases the number of sine tones and harmonics, resulting in a thicker (more) or thinner (less) sound.
- *Percussion Level slider:* Adds a 2nd or 3rd harmonic to the sound, changing both the color and timbre of the instrument.
- *Percussion Time slider:* Sustains the 2nd or 3rd harmonic when set to a long value. If a short value is selected, the harmonics are heard only during the initial keystrike.
- *Click slider:* Introduces a click sound to the keystrike. Select a high level if you'd like this to be clearly heard.
- *Distortion slider:* Makes the sound rough, dirty, and noisy. This is great if you want to do Deep Purple covers!
- *Rotary Speaker buttons:* Choose one of three speaker effects.
  - *Chorale:* Makes the sound swirl.

- *Brake:* Makes the sound swirl initially, and then slow down.
- *Tremolo:* Makes the sound wobble.

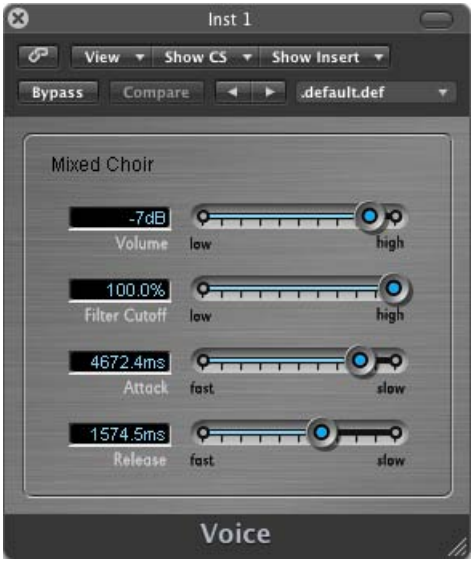
GarageBand Tuned Percussion

The Tuned Percussion sound is sample based. It emulates a vibraphone, xylophone, timpani, steel drums, and other tuned percussion instruments.

- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Voice

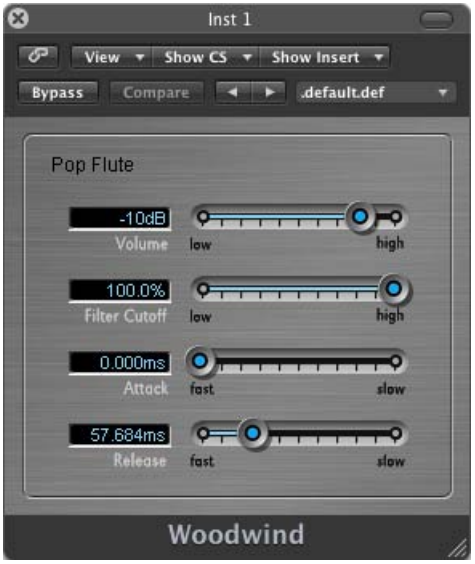
The Voice sound is sample based. It emulates a mixed choir.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

GarageBand Woodwind

The Woodwind sound is sample based. It emulates the sound of wind instruments, such as flutes, clarinets, saxophones, and several other instruments from various world cultures.



- *Volume slider:* Sets the overall volume level of the instrument.
- *Filter Cutoff slider:* Allows more (high) or less (low) of the sound through, making it brighter or damping it.
- *Attack slider:* Makes the sound start more slowly or quickly. A *fast* setting makes it sound like striking a piano key, whereas a *slow* setting makes it sound like bowing a violin string.
- *Release slider:* Determines the time it takes for notes to fade out after you let go of the keys on your keyboard.

Synthesizer Basics

If you are new to synthesizers, read this appendix which discusses important facts about the synthesizer and explains the difference between analog, digital, and virtual analog synthesizers. You will be introduced to all major synthesizer terms and will learn about the basic workings of these hardware- or software-based devices. This appendix is not a detailed, scientific treatise on the inner workings and mathematical theories of synthesis. It is a simple guide to what you need to know, including some extras that are nice to know.

Take your time to read through the appendix as laid out, and feel free to experiment with the ES1, ES2, and other Logic Express instruments. Seeing and using the parameters and other elements onscreen will help you to better understand the conceptual and practical aspects of synthesizers.

This appendix covers the following:

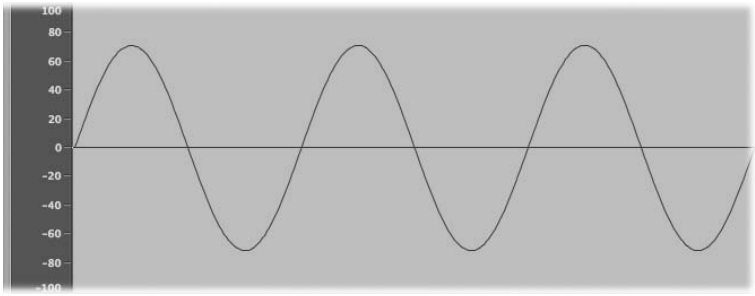
- Sound Basics (p. 349)
- What Is a Synthesizer? (p. 353)
- How Subtractive Synthesizers Work (p. 355)
- Other Synthesis Methods (p. 369)
- A Brief History of the Synthesizer (p. 373)

Sound Basics

Prior to considering any of the sound-generating components you will find in a synthesizer, it's important that you understand sound itself.

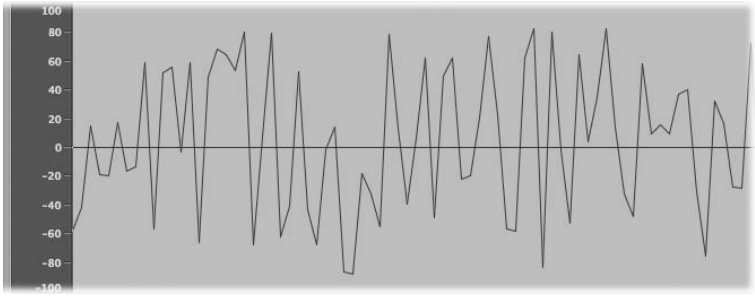
Technically, sound is the conversion of physical energy—such as a hand clap—to an air pressure disturbance. This change in air pressure is transmitted as a series of vibrations—a sound wave—through the air. Sound vibrations can also be transmitted through other matter, such as a wall or floor.

If the vibrations follow a periodic pattern, the sound is said to have a waveform.



The figure above shows an oscillogram—a graphical representation—of a sine wave, the simplest and purest kind of waveform.

If the vibrations do not follow a discernible pattern, the sound is called *noise*.



A repetition of a waveform—each peak and trough in the oscillogram—is known as a *cycle*. The number of cycles that occur per second determines the basic pitch of the waveform—commonly known as the *frequency*. Most Logic Express instruments provide a Hz (Hertz) or Frequency control that determines the number of cycles per second—and therefore the pitch.

Tones, Overtones, Harmonics, and Partials

The frequency of a sound is known as its *fundamental tone*.

The waveforms of all sounds, apart from a basic sine wave, consist of the fundamental tone and many other tones of different frequencies. Nonfundamental tones that are whole-number multiples of the fundamental tone are known as *overtones* or *harmonics*. Nonfundamental tones that are multiplied by fractional amounts (not whole numbers) are called *partials*. A tone divided by the frequency of the fundamental tone is referred to as a *subharmonic*.

- The fundamental tone is referred to as the *first harmonic*. This is generally louder than the other harmonics.

- A tone played at twice the frequency of the first harmonic is called the *second harmonic*.
- A tone played at four times the frequency of the first harmonic is called the *fourth harmonic*, and so on.

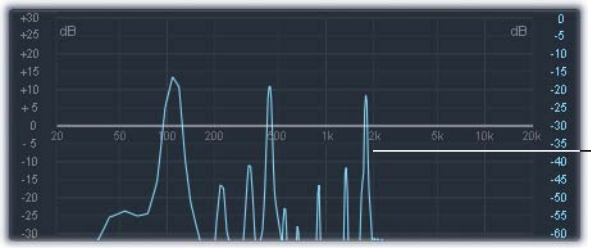
Each of these harmonics has a different timbral quality from the fundamental tone. In general, harmonics that can be multiplied or divided by a whole number–such as octaves, odd-numbered or even-numbered harmonics, and so on–sound more "musical."

Tones that cannot be multiplied or divided by a whole number are known as *inharmonic overtones,* or *partial tones*. When you combine a number of these inharmonic overtones, it tends to sound "noisy."

The Frequency Spectrum

A fundamental tone, when combined with various harmonics of different levels, is perceived as a sound. The level relationships between these sonic elements change over time (controlled by *envelopes*, discussed later in this appendix). The combination of a number of harmonics is referred to as the *harmonic spectrum* or, more commonly, the *frequency spectrum*.

The frequency spectrum shows all the individual sonic elements in a sound. It is shown low to high from left to right. The respective levels of all harmonics are reflected vertically, with taller spikes indicating higher levels.

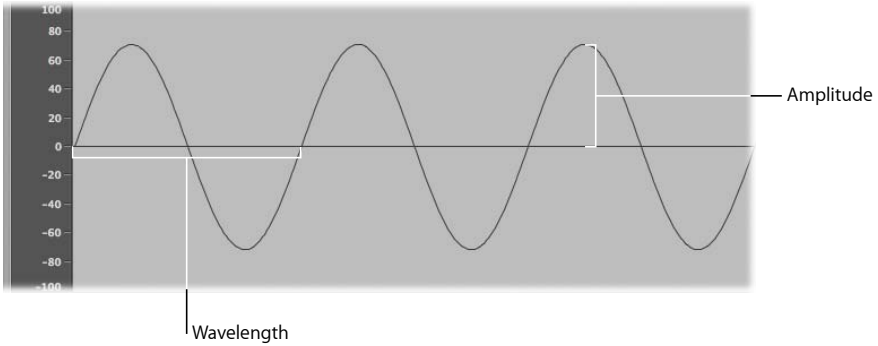


Frequency spectrum graphic of an organ sound

The illustration shows the level and frequency relationships between the fundamental tone and the harmonics at a particular moment in time. These relationships constantly change over time, which results in continuous changes to the frequency spectrum and, therefore, changes to the sound.

Other Waveform Properties

A sound wave, as discussed earlier, has a *frequency*. Other properties of sound waves include *amplitude, wavelength, period,* and *phase*.

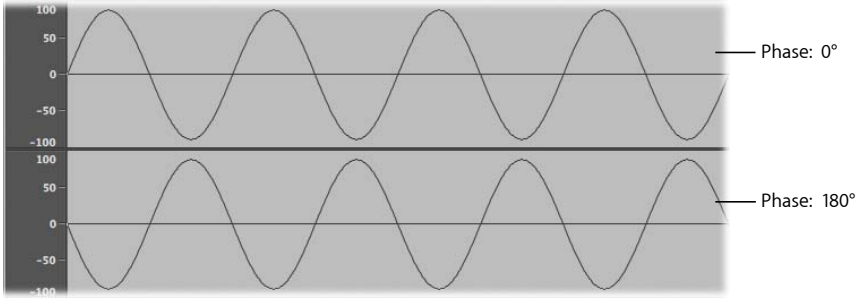


- *Amplitude:* The amplitude of a waveform indicates the amount of air pressure change. It can be measured as the maximum vertical distance from zero air pressure, or "silence" (shown as a horizontal line at 0 dB in the illustration). Put another way, amplitude is the distance between the horizontal axis and the top of the waveform peak, or the bottom of the waveform trough.
- *Wavelength:* The wavelength is the distance between repeating cycles of a waveform of a given frequency. The higher the frequency, the shorter the wavelength.
- *Period:* The (wave) period is the amount of time it takes to complete one full revolution of a waveform cycle. The higher and faster the frequency, the shorter the wave period.
- *Phase:* Phase compares the timing between waveforms and is measured in degrees—from 0 to 360.

When two waveforms begin at the same time, they are said to be *in phase* or *phase aligned*. When a waveform is slightly delayed in comparison to another waveform, the waveforms are said to be *out of phase*.

Note: It is difficult to discern a constant phase difference over the entire wave period, but if the phase of one of the waveforms changes over time, it will become audible. This is what happens in common audio effects such as *flanging* and *phase shifting*.

When you play two otherwise identical sounds out of phase, some frequency components—harmonics—can cancel each other out, thereby producing silence in those areas. This is known as *phase cancellation*, and it occurs where the same frequencies intersect at the same level.



Fourier Theorem and Harmonics

"Every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relationships (ratios of small numbers)." This is known as the Fourier theorem. Roughly translated into more musical terms, this means that any tone with a certain pitch can be regarded as a mix of sine tones. This is comprised of the fundamental tone and its harmonics (overtones). As an example:The basic oscillation (the fundamental tone or first harmonic) is an "A" at 220 Hz. The second harmonic has double the frequency (440 Hz), the third one oscillates three times as fast (660 Hz), the next ones four and five times as fast, and so on.

What Is a Synthesizer?

Sound synthesis is the electronic production of sounds—starting from their basic properties, such as sine tones and other simple waves.

Synthesizers are so named because they can emulate, or *synthesize*, a wide variety of sounds—such as the sound of another instrument, a voice, a helicopter, a car, or a dog bark. Synthesizers can also produce sounds that don't occur in the natural world. The ability to generate tones that cannot be created in any other way makes the synthesizer a unique musical tool.

The simplest form of synthesizer would be a basic sine wave generator that provided little or no control over pitch. Unfortunately, you could not use such a simple synthesizer to synthesize anything except a sine wave.

Combining multiple sine generators with pitch control, however, can produce many interesting and useful tones.

In a synthesizer, the task of tone generation falls to a component known as an *oscillator*. Most synthesizer oscillators generate harmonically rich waveforms, such as *sawtooth, triangle, square* and *pulse* waves. These waves are named due to the resemblance of their shapes to the teeth on the blades of a saw, to triangles, to squares, and so on. For information about the most common synthesizer waveforms, see Oscillators.

Sculpting the fundamental tone and related harmonics into another sound is achieved by routing the signal from one component, also known as a *module*, to another in the synthesizer. Each module performs a different job that affects the source signal.

In a modular synthesizer, this routing is achieved by physically cabling modules to each other. In most modern synthesizers, however, the signal routing between modules is internally prewired and is typically changed using switches, knobs, and other controls.

For a discussion of the various synthesizer components and how they interact with each other to control and shape your sound, see How Subtractive Synthesizers Work.

Synthesizers have been around for much longer than you may think. In the days that preceded the use of digital technology, all electronic synthesizers were analog. And prior to the use of electricity, synthesizers were actually mechanical. If you're interested in learning more, see A Brief History of the Synthesizer.

Analog

An analog synthesizer combines voltage-controlled circuits—such as oscillators, filters, and amplifiers—to generate and shape sounds. The amount of voltage is typically related directly to the waveform pitch, with higher voltages equalling higher pitches.

Digital

In a digital synthesizer, the signal flow is digital. Binary *descriptions* of the signal—a string of zeros and ones—are fed from one algorithm to another.

Hybrid Analog and Digital Synthesizers

Some synthesizer designs feature digital oscillators—which generate signals that are sent to analog filters—and amplifiers. The main advantage of this approach is that digital oscillators don't drift in pitch, a problem that analog oscillators are often notorious for.

Virtual Analog

A virtual analog synthesizer is a digital synthesizer that mimics the architecture, features, and peculiarities of an analog synthesizer. The behaviors and functions of the oscillators, filters, and other modules that you would find in an analog synthesizer are emulated by computer algorithms.

The ES1 is a great example of a virtual analog synthesizer. Its virtual signal flow is that of a typical analog synthesizer, but all the signal processing—the virtual oscillators and so on—is performed by the central processing unit (CPU) of your computer.

The ES1 includes some of the desirable idiosyncrasies of particular analog circuits—in cases where they tend to sound nice, such as high oscillator levels overdriving the filter. And undesirable analog synthesizer phenomena, such as the habit of going completely out of tune, are not simulated.

Virtual analog synthesizers have a few other advantages over their analog counterparts:they're completely programmable (you can save sound settings), they can be completely automated (you can record and play back fader and knob movements), and they are often multi-timbral (you can play different sounds at the same time, on different instrument channels). Aspects such as polyphony (the ability to play multiple notes) and velocity sensitivity are found in most virtual analog synthesizers but in very few analog instruments.

How Subtractive Synthesizers Work

There are a number of approaches to sound creation with a synthesizer (see Other Synthesis Methods). Although there are numerous differences between synthesizer models, most follow a fundamentally similar architecture and signal flow that is based on subtractive synthesis principles.

According to legend, when Michelangelo was asked how he managed to carve David out of a block of stone, he replied, "I just cut away everything that doesn't look like David."

In essence, this is how subtractive synthesis works:You filter (cut away) the parts of the sound that you don't want to hear. In other words, you subtract parts of the frequency spectrum, consisting of the fundamental tone and associated harmonics.

The subtractive approach to synthesis assumes that an acoustic instrument can be approximated with a simple oscillator—which produces waveforms with different frequency spectrums. The signal is sent from the oscillator to a filter that represents the frequency-dependent losses and resonances in the body of the instrument. The filtered (or unfiltered) signal is shaped over time by the amplifier section of the synthesizer.

The distinctive timbre, intonation, and volume characteristics of a real instrument can theoretically be recreated by combining these components in a way that resembles the natural behavior of the instrument you are trying to emulate.

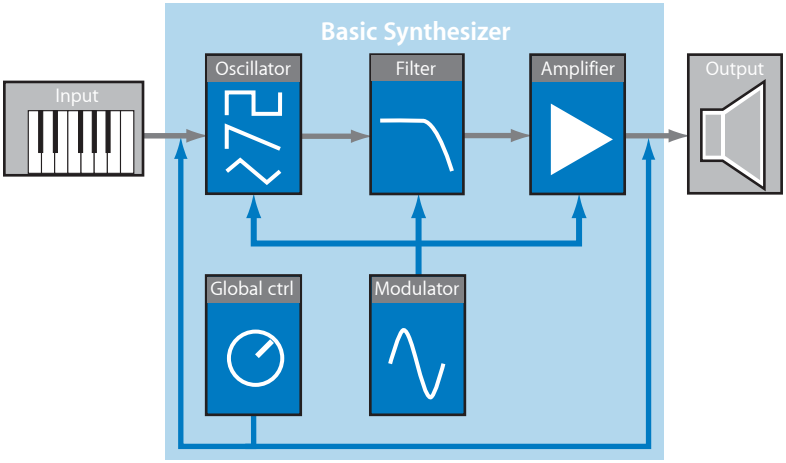
In reality, however, because subtractive synthesizers aren't ideal for emulating real world instruments, no synthesized clarinet is going to fool anyone—particularly now that there are multigigabyte sound libraries available for samplers like the EXS24 mkII.

The real strength of subtractive synthesizers is that they offer a unique sound palette of their own.

All analog and virtual analog synthesizers use subtractive synthesis to generate sound.

An Overview of Subtractive Synthesizer Components

The front panel of most subtractive synthesizers contains a collection of similar signal generating and processing modules—coupled with a number of modulation and control modules. The signal-generating and processing modules typically run left to right, mirroring the actual synthesizer signal flow.



Signal Generating and Processing Components

- *Oscillators:* Generate the basic signal. This is usually a waveform that is rich in harmonics (see Oscillators). Many synthesizers offer more than one oscillator.
- *Filter section:* Used to alter the basic signal by filtering out (removing) portions of the frequency spectrum. Many synthesizers offer a single filter, which is applied universally to all oscillator signals. Multioscillator synthesizers can provide multiple filters, allowing each oscillator signal to be filtered in a different way (see Filters).
- *Amplifier section:* Used to control the level of the signal over time. The amplifier features a module known as an *envelope*, which is broken down into several elements that provide level control for the beginning, middle, and end portions of your sound. Simple synthesizers generally offer a single envelope, which is used to control the oscillator (and filter) over time. More complex synthesizers can feature multiple envelopes (see Envelopes in the Amplifier Section).

Modulation and Control Components

- *Modulators:* Used to modulate the signal-generating and processing components. Modulations can be machine-based—automatically generated by a synthesizer component—or can be manually activated by using the modulation wheel, for example. Most synthesizers provide a component called an *LFO* (low frequency oscillator) to provide a waveform that modulates the signal. See Modulation.

- *Global controls:* Affect the overall characteristics of your synthesizer sound, such as glides between notes, pitch bend, monophonic or polyphonic playback, and more (see Global Controls).

Oscillators

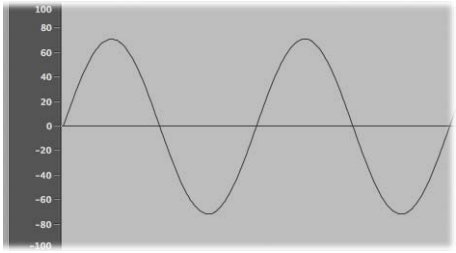
The audio signal of a synthesizer is generated by the oscillator. Usually you would choose from a selection of waveforms that contain differing types and varying amounts (more or fewer) of harmonics. The level relationships between the fundamental tone and the harmonics of the chosen waveform are responsible for the basic sound color or timbre.

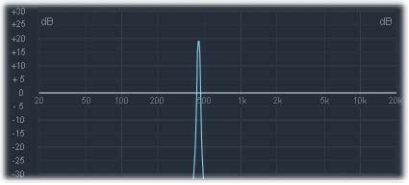
Common Synthesizer Waveforms

The qualities of the most common synthesizer waveforms are discussed below.

Sine Wave

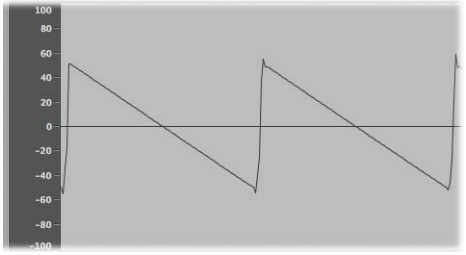
Clean and clear-sounding, a sine wave contains no harmonics but the first harmonic; in other words, it is the fundamental tone. The sine wave—used standalone—can be used to create "pure" sounds like whistles, the sound of wet fingers on the rim of a glass, tuning forks, and so on.

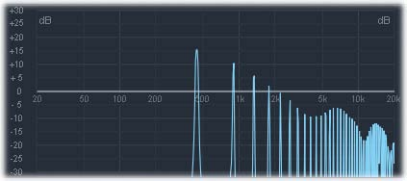




Sawtooth

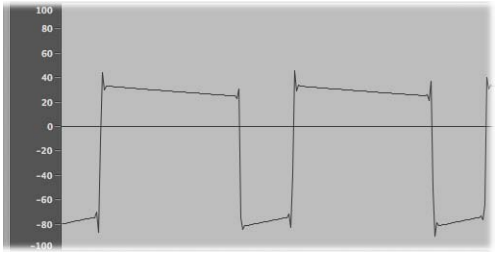
Clear and bright-sounding, a sawtooth wave contains both odd and even harmonics. It is ideal for the creation of string, pad, bass, and brass sounds.

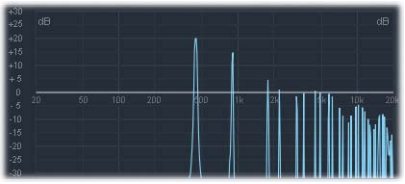




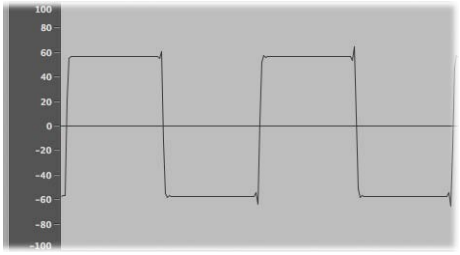
Square and Pulse Waves

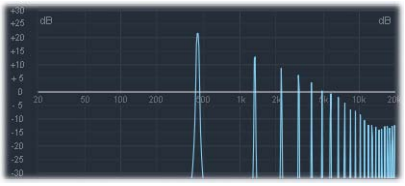
Hollow and woody-sounding, a square wave can contain a wide range of odd harmonics. It is useful when creating reed instruments, pads, and basses. It can also be used to emulate kick drums, congas, tom-toms, and other percussive instruments—often blended with another oscillator waveform, such as noise.





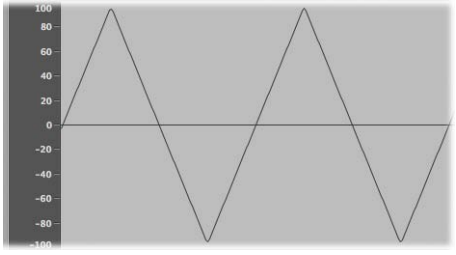
The square wave can be reshaped to make the waveform cycles—or pulses—more rectangular on many synthesizers, using a pulse width modulation (PWM) control. The more rectangular the wave becomes, the more nasal it sounds. When modulated in this way, the square wave is known as a *pulse wave*, and contains fewer harmonics. It can be used for reeds, basses, and brass sounds. See Reshaping Waveforms.

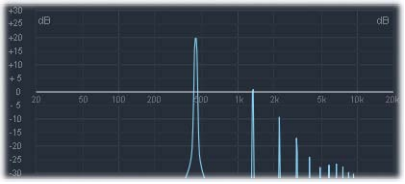




Triangle

Like a square wave, a triangle wave contains only odd harmonics. Because a triangle wave's higher harmonics roll off faster than the ones of a square wave, the triangle wave sounds softer. It is ideal for creating flute sounds, pads, and vocal "oohs."

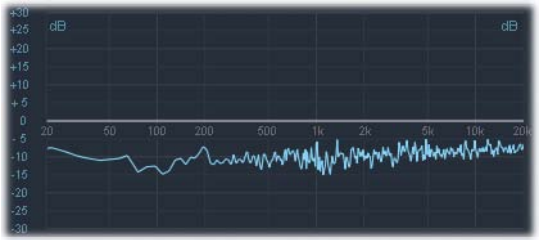




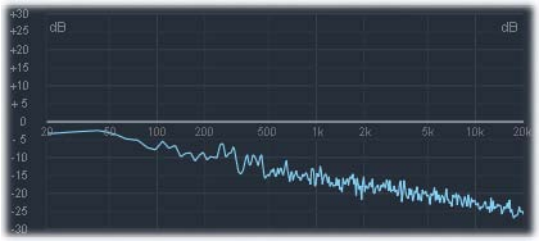
Noise: Pink/Red, Blue, White

Noise is useful for emulating percussive sounds, such as snare drums, or wind and surf sounds, among others.

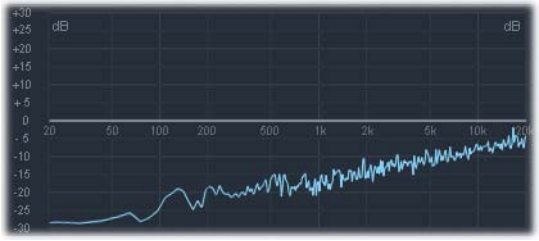
- *White noise:* The most common noise waveform found on synthesizers. White noise contains all frequencies—at full level—around a center frequency.



- *Pink and red noise:* These noise colors also contain all frequencies, but they are not at full level across the frequency spectrum. Pink noise decreases the level by 3 dB per octave (of higher frequencies). Red noise decreases the level by 6 dB per octave.



- *Blue noise:* Blue noise, which is the reverse of pink noise, increases the level of all frequencies in higher octaves by 3 dB.



There are other noise wave colors, but they aren't commonly found in synthesizers.

Reshaping Waveforms

It is possible to deform the basic waveforms to create new waveforms. This results in a different timbre, or tonal color, thus expanding the palette of sounds that can be created.

There are many ways to reshape a waveform. The most obvious would be altering the pulse width of a square wave, as discussed in Common Synthesizer Waveforms. Other waveform-altering options include changing the phase angle, moving the start point of a waveform cycle, or simply combining multiple waveforms in multioscillator synthesizers.

When reshaped in these and other ways, the relationships between the fundamental tone and other harmonics change, thus altering the frequency spectrum and the basic sound being produced.

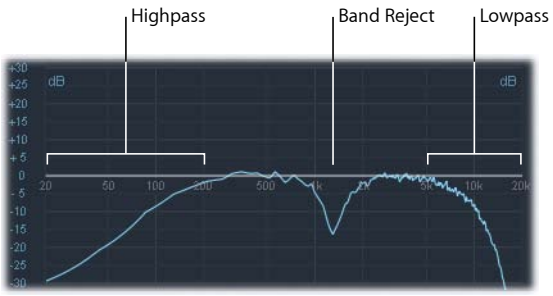
Filters

The purpose of the filter in a subtractive synthesizer is to remove portions of the signal—the frequency spectrum—that is sent from the oscillators. After being filtered, a brilliant-sounding sawtooth wave can become a smooth, warm sound without sharp treble.

The filter sections of most subtractive synthesizers contain two primary controls known as *cutoff frequency*—often simply called *cutoff*—and *resonance*. Other filter parameters can include *drive* and *slope*. The filter section of most synthesizers can be modulated by envelopes, LFOs, the keyboard, or other controls such as the modulation wheel.

Types of Filters

There are several filter types. Each has a different effect on various portions of the frequency spectrum:

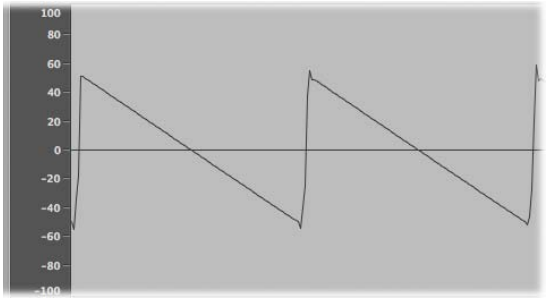


- *Lowpass filter:* Low frequencies are passed; high frequencies are attenuated.
- *Highpass filter:* High frequencies are passed; low frequencies are attenuated.
- *Bandpass filter:* Only frequencies within a frequency band are passed.
- *Band Reject filter:* Only frequencies within a frequency band are attenuated.
- *Allpass filter:* All frequencies in the spectrum are passed, but the phase of the output is modified.

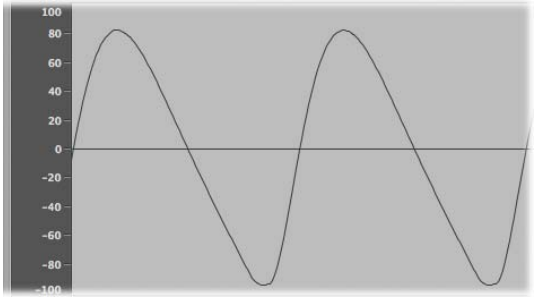
Cutoff Frequency

The cutoff frequency, or *cutoff*, as the name suggests, determines where the signal is cut off. Simpler synthesizers offer only lowpass filters. Thus, if a signal contains frequencies that range from 20 to 4000 Hz, and a cutoff frequency is set at 2500 Hz, frequencies above 2500 Hz are filtered. The lowpass filter allows frequencies below the cutoff point of 2500 Hz to pass through, unaffected.

The figure below shows an overview of a sawtooth wave (A = 220 Hz). The filter is open, with cutoff set to its maximum value. In other words, this waveform is not being filtered.



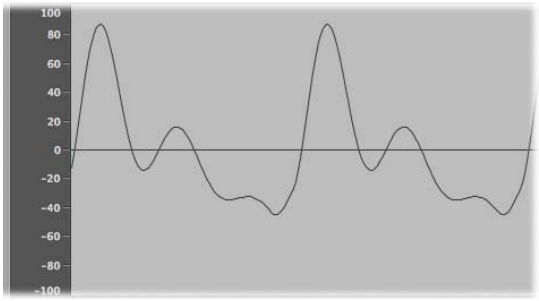
The figure below shows a sawtooth wave with the filter cutoff set to about a 50% value. This filter setting results in suppression of the higher frequencies and a rounding of the edges of the sawtooth wave, making it resemble a sine wave. Sonically, this setting makes the sound much softer and less brassy.



As you can see from this example, use of the filters to cut away portions of the frequency spectrum alters the waveform's shape, thus changing the timbre of the sound.

Resonance

The resonance control emphasizes or suppresses signals around the cutoff frequency. The figure below shows an ES1 sawtooth wave with a high resonance setting and the cutoff frequency set to 660 Hz, which is about 60%.



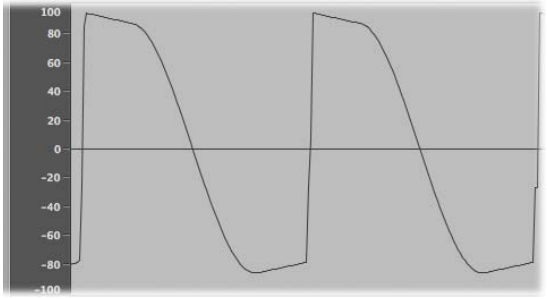
This resonant filter setting results in much brighter and harsher signals close to the *cutoff frequency*. Frequencies below the cutoff point are unaffected.

Once again, the overall result of using filter resonance is a change in the basic waveform's shape and, therefore, its timbre.

Very high filter resonance settings can be used to such an extreme degree that the filter begins to self-oscillate, resulting in the filter generating a sine wave.

Drive

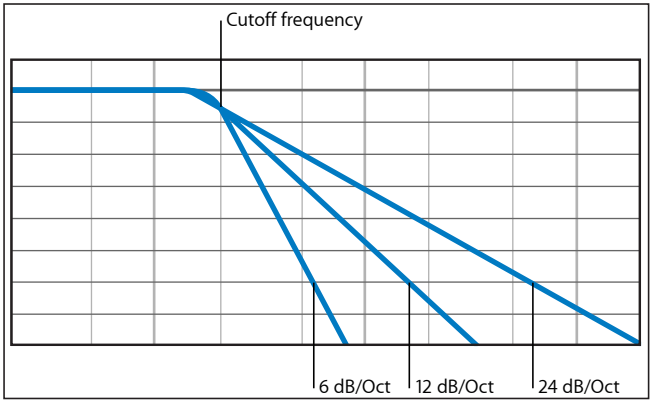
Drive adds an amount of gain to the waveform as it enters the filter—an input gain control—thus overdriving the filter and distorting the waveform. This waveform distortion changes the timbre of the sound, making it much harsher. See Reshaping Waveforms for more information about waveform distortions.



The figure shows an unfiltered sawtooth wave, with Drive set to about 80%. Notice that the wave cycles touch the floor and ceiling of the filter's dynamic range.

Filter Slope

As discussed earlier, a filter will cut off the signal at the set cutoff frequency. This cutoff doesn't happen abruptly but rather at a given slope, which is measured in decibels (dB) of gain reduction per octave. Put another way, you can define how steep the "cliff" is at the cutoff point by choosing a relatively severe or more gentle slope.



Envelopes in the Amplifier Section

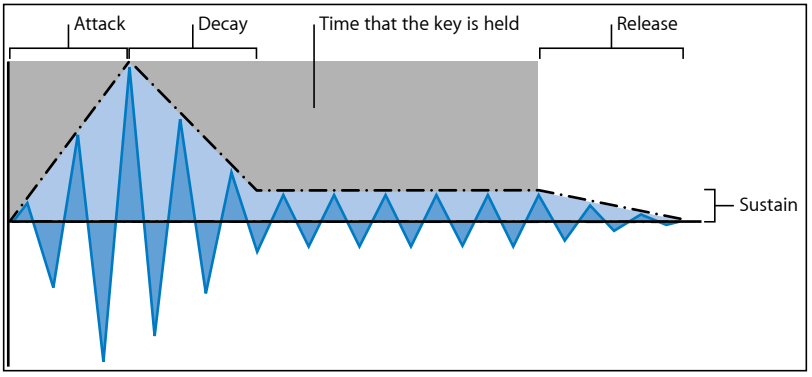
The amplifier module of a synthesizer is responsible for controlling the level—or loudness—of the signal over time.

To put this into a musical context, consider the sound of a violin, which slowly ramps up to a peak—or maximum—level as the bow is smoothly dragged across a string, is sustained for a period until the bow is moved away from the string, and then cuts off abruptly. In comparison, hitting a snare drum with a drumstick results in a very fast peak level, with no sustain portion, and the sound immediately dies out (although there will be an amount of *decay*, the time it takes to fall from the peak level). As you can see, these two sounds have very different characteristics over time.

Synthesizers emulate these sonic characteristics by providing control over different parts—the beginning, middle, and end—of a sound's level over time. This control is achieved with a component called an *envelope generator*.

The Attack, Decay, Sustain, and Release (ADSR) Envelope Controls

An oscillogram of a percussive tone is shown below in which the level rises immediately to the top of its range and then decays. If you drew a box around the upper half of the oscillogram, you could consider it the "envelope" of the sound—an image of the level as a function of time. It is the function of the envelope generator to set the shape of this envelope.



The envelope generator usually features four controls—attack, decay, sustain and release, commonly abbreviated as ADSR.

- *Attack:* Controls the time it takes for the initial slide from an amplitude of zero to 100% (full amplitude).
- *Decay:* Determines the time taken for the subsequent fall from 100% amplitude to the designated sustain level.
- *Sustain:* Sets the steady amplitude level produced when a key is held down.
- *Release:* Sets the time it takes for the sound to decay from the sustain level to an amplitude of zero when the key is released.

If a key is released during the attack or decay stage, the sustain phase is usually skipped. A sustain level of zero will produce a piano-like—or percussive—envelope, with no continuous steady level, even when a key is held.

Using the Envelope to Control Filters

Envelope generators are not limited to controlling signal amplitude. They can also control the rise and fall of the filter cutoff frequency or modulate other parameters. In other words, envelope generators can be used as a modulation source—or as a "remote control" for a given parameter, if you prefer.

This aspect of synthesizers—modulation—is covered in the following section.

Modulation

Without modulation, sounds tend to be boring and fatiguing to the ear. They also sound synthetic, rather than natural, in the absence of some type of sonic modulation. The most obvious type of modulation is vibrato, which is used by orchestral string players to add animation to an instrument's pitch.

To make sounds more interesting, you can use various synthesizer controls to modulate basic sound parameters.

Modulation Routing

Many synthesizers, including the ES1, ES2, and EXS24 mkII sampler, feature a modulation router.

The router allows you to direct one or more modulation sources to one or more modulation targets—or destinations, if you prefer. For example, you can change modulation targets, such as the oscillator pitch or filter cutoff frequency, by using modulation sources that include the following:

- *Velocity modulation:* The impact of your keyboard playing (harder or softer).
- *Key scaling:* The position you play on the keyboard.
- *Use of controls:* These can include the modulation wheel, ribbon controllers, or pedals attached to your keyboard.
- *Automatic modulation:* You can use envelope generators or LFOs to modulate signals automatically.

Modulation Routing in the ES1 and ES2

The ES1 and ES2 provide an easy way to route a control—a modulation source—to part of the sound engine—a modulation target. For details on the use of the modulation facilities and other parameters, see the ES1 and Oscillator on/off buttons.

You can do modulation routing n the ES1 by selecting a modulation target in the left or right column of buttons in the Router section. You use the left column to set a modulation target that can be controlled, in amount, via the modulation wheel of your keyboard. The target you choose in the right column will dynamically respond to keyboard velocity. The amount, or range, of this modulation is determined by the two arrows shown in the sliders (Int via Whl and Int via Vel). The upper arrow determines the maximum amount of modulation, and the lower arrow determines the minimum amount of modulation.

The ES2 provides ten modulation routings—in columns. Although it looks somewhat intimidating at first, each routing column is quite similar to the modulation controls found in the ES1. Note the first routing, at the left in the figure below:

The modulation target is Pitch123. The pitch—the Frequency parameter—of oscillators one, two, and three is affected (by LFO2, the modulation source).

LFO2 is the modulation source. The two arrows to the right of the column indicate the modulation amount. To make the modulation more intense, drag the upper and/or lower arrows up or down, thereby increasing the range of the modulation amount. The upper arrow determines the maximum amount of modulation, and the lower arrow determines the minimum amount of modulation.

The Via control is the ModWhl. The amount of modulation—the range of which is determined by the sliders to the right of the channel—is directly controlled via the modulation wheel of your keyboard. When the modulation wheel is at the minimum setting, at the bottom of its travel, the amount of oscillator pitch modulation is minimal—or off/no modulation at all. As you move the modulation wheel upward, the frequency of all three oscillators is directly controlled by the LFO within the range determined by the sliders.

Common Modulation Sources

This section outlines modulation sources that you typically find on most synthesizers.

Modulation Controllers

Modulation sources can be—and often are—triggered by something you've done, such as played a note on the keyboard, or moved the modulation wheel.

Thus, the modulation wheel, pitch bend ribbons, foot pedals, keyboard, and other input options are referred to as *modulation controllers*, or simply *controllers*.

Perhaps the best example of a modulation controller in action would be the use of a velocity-sensitive keyboard, set up to control the filter and level envelopes. The harder you strike the notes, the louder and brighter the sound is. See Using Envelopes for Modulation.

Using the LFO to Modulate Sounds

A modulation source found on nearly all synthesizers is the LFO (low frequency oscillator). This oscillator is used only as a modulation source and does not generate any audible signals that form part of your actual synthesizer sound, because it's too low to be heard. It can, however, affect the main signal by adding vibrato, filter sweeps, and so on.

LFO Controls

The LFO generally offers the following controls:

- *Waveform:* Allows you to choose the type of waveform (triangle waves and square waves are seen most often). Triangle waves are useful for filter sweeps (slow changes to the filter cutoff frequency) or when simulating an ambulance siren (slow changes to the oscillator frequency). The square waveform is useful for rapid switches between two different pitches (vibratos or octaving, for example).
- *Frequency/Rate:* Determines the speed of the waveform cycles produced by the LFO. When set to low values, very slow ramps are produced, making it easy to create sounds such as ocean waves rolling in (when white noise is chosen as the waveform in the main oscillator).
- *Sync mode:* Allows you to choose between free running (a user-defined LFO rate) or synchronization with an external tempo source (such as a host application).

LFO Envelopes

The LFO can also be controlled with an envelope generator in some synthesizers. As an example of where this might be useful, imagine a sustained string section sound where vibrato is introduced a second or so into the sustained portion of the sound. If this can happen automatically, it allows you to keep both hands on the keyboard.

In certain synthesizers, a simple envelope generator is included for this precise purpose. Often, this envelope consists only of an attack parameter, or occasionally it includes decay or release options. These parameters perform in the same fashion as the amplitude envelope parameters (see Envelopes in the Amplifier Section), but they are limited to control of LFO modulations.

Using Envelopes for Modulation

The main envelope generator of the synthesizer not only controls levels over time, but it also is often used to modulate other sound parameters when you press or release keyboard keys.

The most common use of envelope modulation is to control the filter cutoff and resonance parameters with the keyboard velocity or keyboard scaling modulation sources (see Modulation Routing).

Global Controls

This section covers global controls that affect the overall output signal of your synthesizer.

The most obvious global control is the Level control, which sets the overall loudness of your sound. For more information about the Level control, see Envelopes in the Amplifier Section.

Other key global controls include the following:

- *Glide (sometimes called Portamento):* Used to set the amount of time that it takes for one note pitch to slide up or down to another note pitch. This control is useful for emulating wind instruments that slide from note to note, rather than move directly to another clear and distinct pitch.
- *Bender/Bend Range:* This control is generally hard-wired to a Pitch Bend wheel on a keyboard. As the name suggests, moving the wheel up or down from its centered position bends the pitch (the oscillator frequency) up or down. The Bender/Bend Range parameter usually has an upper and lower limit of one octave but is typically set to around three semitones up or down. This setting is ideal for emulating small (or extreme) pitch fluctuations that occur in some instruments—such as when moving between notes with a trumpet, or bending the strings during a scorching guitar solo.
- *Voices:* Synthesizers have a limit to the number of notes that can be produced simultaneously. Producing notes simultaneously is known as the *polyphony* (literally, this means "many voices") of the instrument. The Voices parameter sets an upper limit to the number of notes that can be played at a given time.
- *Unison:* Used to "stack" voices—with the unison voice being heard one octave above the frequency of the played note. Because two voices are being used when you play a note, unison has two effects—it makes the sound richer and fuller, and it halves the polyphony.
- *Trigger Mode:* Trigger mode determines how the polyphony of the instrument is handled when the number of notes played exceeds the number of available voices. Trigger Mode also allows you to assign legato mode. Essentially, this control changes the way the synthesizer responds to your playing technique and is invaluable when you are emulating monophonic instruments, such as flutes, clarinets, and trumpets. When you use the Trigger Mode control, if you assign a last note priority, a playing note will be cut off by playing another note.
  - *Last note priority:* When new notes are triggered while all voices are playing, the synthesizer frees up polyphony (voices) by ending the notes played earliest. This is the default trigger mode of Logic Express synthesizers when in a monophonic mode.
  - *First note priority:* Notes played earlier are not stopped. In this mode you need to stop playing notes in order to play a new one after you have reached the limit of the polyphony (voices) of the instrument.

Note: The Trigger Mode parameter can also allow you to set priorities for lower- or higher-pitched notes when playing monophonically (one voice at a time) in some synthesizer designs.

There are many other global controls found on different synthesizer models that have an impact on your overall sound.

Other Synthesis Methods

There are many ways to create sounds, using different technologies and approaches to synthesis. This section covers all the main methods, with reference to Logic Express instruments where applicable.

Many of the outlined methods incorporate into their design at least some elements of the subtractive synthesis approach covered earlier. The most common modern approach is based on samples of real instruments and sounds.

Sample-Based Synthesis

Sample-based synthesis, which is sometimes known as *Pulse Code Modulation* (PCM), or *sampling and synthesis* (S&S) synthesis, is differentiated from subtractive synthesis mainly by the use of samples in place of oscillator waveforms.

The samples—digital recordings of existing sounds—are mapped across the keyboard. Typically, each sample is mapped to a note in the center of a keyboard range that spans 5 or so notes that are unique to that sample. The reason for this range of 5 or so notes is that samples tend to sound much less like the source sound if played more than a few notes higher or lower than the original pitch—due to the relationship between the pitch and playback speed of samples.

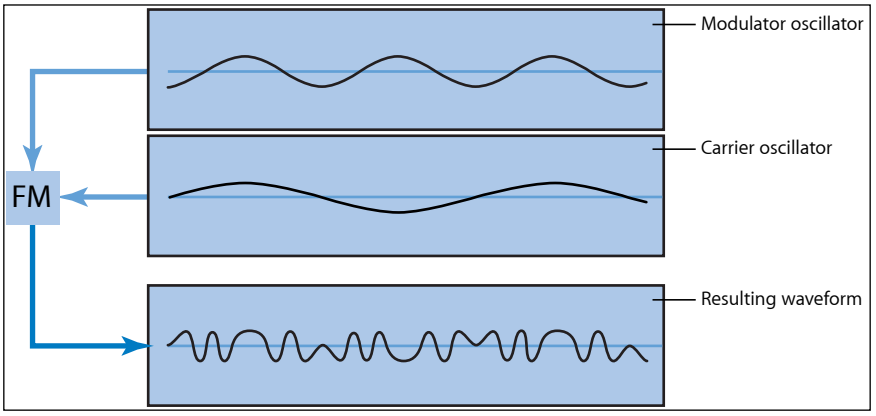
The pitch of each sample isn't changed with a frequency control, unlike the oscillator waveform of a synthesizer that is not sample based. Rather, a sample is played back at a faster or slower speed to alter its pitch, which has a corresponding impact on the sample playback time. For example, a sample played back at twice the speed requires half the time to play through.

The EXS24 mkII is a sample player that can be used much like a sample-based synthesizer, due to the subtractive synthesis facilities that it offers.

Popular instruments that use this synthesis approach include Korg's M1, O1/W, and Triton; the Roland JV/XP instruments; Yamaha's Motif series; and many others.

Frequency Modulation (FM) Synthesis

Put simply, FM synthesis involves the use of a modulator oscillator and a sine wave *carrier* oscillator. The modulator oscillator modulates the frequency of the carrier oscillator within the audio range, thus producing new harmonics. These harmonics are known as *sidebands*.



Typically, FM synthesizers don't incorporate a filter. You can generate some subtractive synthesizer style sounds with FM synthesis, but it is difficult to recreate the sound of a resonant subtractive synthesizer filter with this method. FM synthesis is extremely good, however, at creating sounds that are difficult to achieve with subtractive synthesizers-sounds such as bell timbres, metallic tones, and the tine tones of electric pianos. Another strength of FM synthesis is punchy bass and synthetic brass sounds.

Logic Express includes a simple FM synthesizer, the EFM1. Although it is minimalist, it is capable of producing many of the classic FM sounds made famous by Yamaha's DX series of synthesizers (the DX7, sold from 1983 to 1986, remains the most commercially successful professional-level hardware synthesizer ever made).

The ES2 also features some FM techniques that allow you to modulate one oscillator with another. You can use these FM techniques to partially bridge the gap between the very digital sound of FM synthesis and the fat analog sound that the ES2 is noted for.

Component Modeling Synthesis

Also known as *physical modeling,* this synthesis method uses mathematical models to simulate instruments. Parameters are used to describe an instrument's physical characteristics, such as the materials the instrument is made of, the dimensions of the instrument, and the environment it is played in—under water, in air. Equally important are descriptions of how the player would interact with the instrument—for example, whether it is played by plucking, bowing, or strumming strings; by hitting it with sticks; by placing fingers on sound holes; and so on.

To model a drum sound, for example, the following aspects would need to be taken into account. Of primary importance would be the actual drum strike—how hard it is and whether the drumhead is struck with a wooden stick, a mallet, a beater, and so on. The properties of the drumhead (the skin or membrane) would include the kind of material, its degree of stiffness, its density, its diameter, and the way it is attached to the shell of the drum. The volume of the drum cylinder itself, its material, and the resonance characteristics of all of the above would need to be mathematically described.

To model a violin, you'd need to take into account the bow against the string, the bow-width and material, the bow tension, the string material, the string density, the string tension, the resonance and damping behavior of the strings, the transfer of string vibrations through the bridge (materials, size, and shape of the bridge), and the materials, size and resonance characteristics of the violin body. Further considerations would include the environment that your modeled violin is played in, and the playing style-"hammering" or tapping with the bow as opposed to drawing it across the strings.

Logic Express includes Ultrabeat, a drum synthesizer that incorporates physical modelling components.

Wavetable, Vector, and Linear Arithmetic (LA) Synthesis

Wavetable synthesis uses a number of different single-cycle waveforms, laid out in what is known as a *wavetable*.

Playing a note on the keyboard triggers a predetermined sequence of waves. In general, this is not a stepped transition but rather a smooth blend from one waveform into another, resulting in a constantly evolving waveform. Multiple wavetables can also be used simultaneously—either played one after the other, or blended together—resulting in more harmonically complex waveforms.

A single wavetable can emulate filter cutoff with a series of bright, less bright, then dull-sounding waveforms played in sequence—which resembles a reduction of the filter cutoff frequency in a subtractive synthesizer.

Wavetable synthesis isn't particularly successful at emulating acoustic instruments. It is, however, extremely successful at producing constantly evolving sounds; harsh and metallic, or bell-like sounds; punchy basses; and other digital tones.

Wavetable synthesis was championed by the PPG and Waldorf instruments. The ES2 also includes wavetable facilities.

Roland LA (Linear Arithmetic) synthesizers such as the D-50 work on a similar principle. In these synthesizers, however, complex sampled attack phases are combined with simple sustain or decay phases to create a sound. In essence, this is a simple wavetable that consists of two samples.

Where LA and wavetable synthesizers differ is that the latter were designed to create new, original, digital sounds. LA synthesizer designers, in contrast, wanted to emulate real instruments using a minimum of memory. To facilitate this, they combined samples of the attack phase—the crucial part of a sound—with appropriate decay and sustain phases.

Vector synthesis—used in the Sequential Circuits Prophet-VS and Korg's Wavestation—allows you to move through wavetables and sequences arranged on a two-dimensional grid (two different vectors, or less technically, on the X or Y axis). The main benefit of this approach is that the balance between samples and waves is achieved in real time by moving a joystick. You can also use the ES2 to perform vector synthesis by modulating the Oscillator Mix (Triangle) parameter with the Vector Envelope.

Additive Synthesis

Additive synthesis could be considered the reverse approach to subtractive synthesis. See the beginning of this appendix, including the discussion about all sounds being a sum of various sine tones and harmonics, for background information to provide insight into additive synthesis.

In essence, you start out with nothing and then build up a sound by combining multiple sine waves of differing levels and frequencies. As more sine waves are combined, they begin to generate additional harmonics. In most additive synthesizers, each set of sine waves is viewed and used much like an oscillator.

Depending on the sophistication of the additive synthesizer you are using, you will be provided with individual envelope control over each sine wave, or you will be limited to envelope control over groups of sine waves—one envelope per sound and its harmonics, or all odd or all even harmonics, for example.

Logic Express does not contain any instruments that use additive synthesis methods.

Resynthesis

You can analyze the frequency components of a recorded sound and then resynthesize (reconstruct) a representation of the sound using additive techniques. By calculating the frequency and amplitude of each harmonic in the overall frequency spectrum of the sound, an additive resynthesis system can generate a series of sine waves (with appropriate levels over time) for each harmonic.

After the sound has been resynthesized in this fashion, you can adjust the frequency and amplitude of any harmonic. Theoretically, you could restructure a harmonic sound to make it inharmonic, for example.

Phase Distortion Synthesis

Phase distortion synthesis creates different waveforms by modifying the phase angle of a sine wave.

In essence, you can bend a sine wave until it becomes a sawtooth wave, a triangle wave, a square wave, and so on. The synthesizer engine beyond the waveform generation in general follows the standard subtractive method.

Phase distortion synthesis was commercially introduced in 1984's Casio CZ series synthesizers.

Granular Synthesis

The basic premise behind granular synthesis is that a sound can be broken down into tiny particles, or grains. These sampled grains—usually no more than 10 to 50 ms long—can then be reorganized, or combined with grains from other sounds, to create new timbres.

In many respects, this is much like wavetable synthesis, but it works on a much finer scale. As you might expect, this method is ideal for creating constantly evolving sounds and truly unique tones.

The downside is that granular synthesis is extremely processor-intensive, and it wasn't possible to do it in real time until relatively recently. For this reason, it has remained largely ignored by all but a few in academic institutions. Today's computers, however, have sufficient processing power to make this synthesis method a practicality, and there are a number of commercial products now available.

A Brief History of the Synthesizer

This overview showcases a few of the most significant developments in synthesizer history.

Precursors to the Synthesizer

It may surprise you to learn that the earliest seeds of modern electronic synthesizers began in the twilight years of the 19th century. In 1896/1897, an American inventor named Thaddeus Cahill applied for a patent to protect the principle behind an instrument known as the Telharmonium, or Dynamophone. Weighing in at a staggering 200 tons, this mammoth electronic instrument was driven by 12 steam-powered electromagnetic generators. This behemoth was played in real time using velocity-sensitive keys and, amazingly, was able to generate several different sounds simultaneously. The Telharmonium was presented to the public in a series of "concerts" held in 1906. Christened "Telharmony," this music was piped into the public telephone network, because no public address systems were available at the time.

In 1919, Russian inventor Leon Theremin took a markedly different approach. Named after the man who masterminded it, the monophonic Theremin was played without actually touching the instrument. It gauged the proximity of the player's hands, as they were waved about in an electrostatic field between two antennae, and used this information to generate sound. This unorthodox technique made the Theremin enormously difficult to play. Its eerie, spine-tingling (but almost unvarying) timbre made it a favorite on countless horror movie soundtracks. Incidentally, R. A. Moog, whose synthesizers would later garner worldwide fame, began to build Theremins at the tender age of 19.

In Europe, the Frenchman Maurice Martenot devised the monophonic Ondes Martenot in 1928. The sound generation method of this instrument was akin to that of the Theremin, but in its earliest incarnation it was played by pulling a wire back and forth.

In Berlin during the 1930s, Friedrich Trautwein and Oskar Sala worked on the Trautonium, an instrument that was played by pressing a steel wire onto a bar. Depending on the player's preference, it enabled infinitely variable pitches—much like a fretless stringed instrument—or incremental pitches similar to that of a keyboard instrument. Sala continued to develop the instrument throughout his life, an effort culminating in the two-voice Mixturtrautonium in 1952. He scored numerous industrial films, as well as the entire soundtrack of Alfred Hitchcock's masterpiece "The Birds," with this instrument. Although the movie does not feature a conventional musical soundtrack, all bird calls and the sound of beating wings heard in the movie were generated on the Mixturtrautonium.

In Canada, Hugh Le Caine began to develop his Electronic Sackbut in 1945. The design of this monophonic instrument resembled that of a synthesizer, but it featured an enormously expressive keyboard, which responded not only to key velocity and pressure but also to lateral motion.

The instruments discussed thus far were all designed to be played in real time. Relatively early, however, people began to develop instruments that combined electronic sound generators and sequencers. The first instrument of this kind was presented by the French duo Edouard Coupleux and Joseph Givelet in 1929—the inspirationally named Automatically Operating Musical Instrument of the Electric Oscillation Type. This hybrid married electronic sound generation to a mechanically punched tape control. Generally acknowledged to be a mouthful, its unofficial name was shortened to Coupleux-Givelet Synthesizer by its builders; this was, incidentally, the first time a musical instrument was called a "synthesizer."

The term was officially introduced in 1956 with the debut of the RCA Electronic Music Synthesizer Mark I, developed by American engineers Harry F. Olson and Herbert Belar. Its dual-voice sound generation system consisted of 12 tuning forks, which were stimulated electromagnetically. For its time, the instrument offered relatively sophisticated signal-processing options. The output signal of the sound generator could be monitored by loudspeakers and, amazingly, recorded directly onto two records! A single motor powered both turntables and the control unit of the Mark 1. The synthesizer was controlled by information punched onto a roll of paper tape, which actually enabled continuous automation of pitch, volume, timbre, and envelopes. It was as complicated as it sounds-handling was anything but a dream, and spontaneous playing was impossible.

The First Voltage-Controlled Synthesizers

With the exception of the Telharmonium, which was conceived prior to the advent of the thermionic valve, these precursors to the modern-day synthesizer were all based on tube circuitry. This made these instruments relatively unwieldy and certainly volatile. After the transistor became available in 1947/48, more rugged, smaller, and thus portable, instruments were soon to come.

At the end of 1963, American innovator R. A. (Bob) Moog met the composer Herbert Deutsch, who inspired Moog to combine a voltage-controlled oscillator and amplifier module with a keyboard in 1964—the first prototype of a voltage-controlled synthesizer. This collaboration with the German musician prompted Moog to extend his range of modules and to combine them into entire systems. It wasn't until 1967, however, that Moog actually called his diverse mix-and-match systems *synthesizers*.

Moog's achievements spread by word of mouth, and Moog, always keen to elicit the feedback of his customers, continued to add further modules to his line. Wendy Carlos' LP release "Switched-On Bach" (1968) was responsible for the breakthrough of Moog's instruments. The record featured Moog's modular synthesizers and was one of the earliest commercial multitrack recordings. The album's success introduced the synthesizer to a wider audience and made the name Moog synonymous with the instrument. Hoping to capitalize on the new sounds that synthesizers made available, and match Carlos' commercial success, numerous studios, producers, and musicians acquired Moog modular synthesizers. In 1969, as many as 42 employees produced two to three complete modular systems every week at Moog's production facility.

Working independently, an engineer named Donald Buchla had conceived and implemented the concept for a modular, voltage-controlled synthesizer. This coincided with Moog's version. Buchla also developed his first instruments in close cooperation with users. The inspiration for his first synthesizer originated with composers Morton Subotnik and Ramon Sender, of the San Francisco Tape Music Center. Although he began working on this instrument in 1963, it didn't make its public debut until 1966. By design, Buchla's instruments catered primarily to academia and avant-garde musicians, so they never garnered the public attention and acclaim of Moog's synthesizers.

Compact and Cheap

These early voltage-controlled synthesizers were modular.One (or several) chassis housed the power supply and the actual modules. The inputs and outputs of the modules had to be interconnected via a confusing tangle of patch cords before the synthesizer would actually make a sound. Establishing these connections properly was an art unto itself, and obtaining useful settings on the modules required significant expertise.

Moog realized that these modular synthesizers were too complex and expensive for the average musician and were likely to fail if sold through traditional music retailers. In 1969, Moog collaborated with engineers Jim Scott, Bill Hemsath, and Chad Hunt to design a compact, portable, affordable, and easy-to-use synthesizer. After three prototypes were built, the Minimoog Model D was released in the summer of 1970.

Unlike previous modular synthesizers, it was neither necessary (nor possible) for players to connect the modules of the Minimoog as they saw fit. All of the modules' connecting circuitry was hard-wired at the factory. The type and number of modules was also fixed. This simplified manufacturing considerably, and cut costs dramatically. Hard on the heels of a major marketing campaign, the Minimoog became a huge success. Without alteration in its basic design, 13,000 Minimoogs were sold worldwide, right up to 1981.

Storage and Polyphony

Customers weren't entirely satisfied, however. Although musicians no longer had to contend with countless cords in order to play a synthesizer, they still had to deal with loads of knobs and switches before they could do something as simple as switch from one sound to another. Moreover, keyboardists were bored with playing monophonic melody lines on synthesizers—they wanted to be able to play chords. Although dual-voice keyboards that connected two monophonic synthesizers were available as early as 1970, customers wanted more.

Attempting to satisfy these demands, two schools of thought emerged in synthesizer design. One approach called for an independent, monophonic synthesizer to be assigned to every key on the keyboard. To this end, designers married the design principles of electronic organs to synthesizer technology. Although this breed of instrument was fully polyphonic—all notes of the keyboard could be heard simultaneously—it wasn't as versatile in its control options as a true synthesizer. The first fully polyphonic synthesizer to feature this type of design was the Moog Polymoog, released in 1975. Developed primarily by David Luce, it featured 71 weighted, velocity-sensitive keys.

In the second approach to polyphonic sound generation, a synthesizer was assigned to a key only when the key was pressed—in effect, semi-polyphony. As early as 1973, American company E-MU Systems introduced the Modular Keyboard System Series 4050, a digital keyboard that could be connected to up to ten monophonic synthesizers, and thus had ten-voice polyphony. The problems with this approach are obvious—very few people owned ten synthesizers, and the amount of time and effort involved in programming the settings for a new sound were an overwhelming deterrent. Digital memory was still waiting to be developed and, once again, the evolution of semi-polyphonic synthesizers required the desirable qualities that only digital keyboards could provide.

The same prerequisite-digital engineering—eventually led to synthesizers that allowed sounds to be stored. Without the benefit of digital technology, early attempts at storing sounds included some unwieldy solutions. As an example, a synthesizer with analog programmability required a dedicated row featuring all of the instrument's control elements, for every "memory" slot! In this case, a selector switch accessed one of the many identical control panels and connected it to the sound generator.

The first synthesizer featuring storage slots implemented in this manner was the GX1, which Yamaha released in 1975. The control elements for the system's storage slots were so small that they could only be adjusted using jeweller's screwdrivers and complicated tools—called programmers and comparators.

It was not until 1978 that the problem was resolved satisfactorily. The five-voice polyphonic Prophet-5, released by American company Sequential Circuits, was the world's first synthesizer with a global storage facility. All settings for each of its five onboard monophonic synthesizers were stored in memory slots—40 in the debut model. Moreover, all five synthesizers shared a single user interface, which simplified matters considerably. In spite of its initially steep price, this instrument proved extremely popular and approximately 8,000 were built, up until 1985. In addition to its digitally implemented polyphony and memory, the success of the Prophet-5 is attributable to the outstanding quality of its analog sound generation system.

Digital Synthesizers

Even modern digital synthesizers featuring variable polyphony, memory, and completely digital sound generation systems follow this semi-polyphonic approach. The number of voices that these instruments are able to generate, however, is no longer dependent on the number of built-in monophonic synthesizers. Rather, polyphony depends entirely on the performance capability of the computers that power them.

The breathtaking developments in the digital world are best illustrated by the following example. The first program that emulated sound generation entirely by means of a computer was Music I, authored by the American programmer Max Mathew. Invented in 1957, it ran on a university mainframe, an exorbitantly expensive IBM 704. Unimpressively, its sole claim to fame was that it could compute a triangle wave, although doing it in real time was beyond its abilities.

This lack of capacity for real-time performance is the reason why early digital technology was used solely for control (and storage) purposes in commercial synthesizers. Digital control circuitry debuted in 1971 in the form of the digital sequencer found in the Synthi 100 modular synthesizer—in all other respects an analog synthesizer—from English company EMS. Priced out of reach of all but the wealthiest musicians, the Synthi 100 sequencer featured a whopping total of 256 events.

Ever-increasing processor performance made it possible to integrate digital technology into parts of the sound generation engine itself. The monophonic Harmonic Synthesizer, manufactured by Rocky Mountain Instruments (RMI), was the first instrument to do so. This synthesizer had two digital oscillators, combined with analog filters and amplifier circuits.

The Synclavier, introduced in 1976 by New England Digital Corporation (NED), was the first synthesizer with completely digital sound generation. Instruments like the Synclavier were based on specialized processors, which had to be developed by the manufacturers themselves. This development cost made the Synclavier an investment that few could afford.

An alternative solution was the use of general-purpose processors made by third-party computer processor manufacturers. These processors, especially designed for multiplication and accumulation operations—common in audio processing tasks—are called *digital signal processors* (DSPs). Peavey's DPM-3, released in 1990, was the first commercially available synthesizer completely based on standard DSPs. The instrument was 16-note polyphonic and based mainly on three Motorola 56001 DSPs. It featured an integrated sequencer and sample-based subtractive synthesis, with factory presets and user-definable samples.

Another solution was to design synthesizers as a computer peripheral, rather than as a standalone unit. The growing popularity of personal computers from the early 1980s made this option commercially viable. Passport Soundchaser and the Syntauri alphaSyntauri were the first examples of this concept. Both systems consisted of a processor card with a standard musical keyboard attached to it. The processor card was inserted into an Apple II computer. The synthesizers were programmed via the Apple keyboard and monitor. They were polyphonic, had programmable waveforms, envelopes, and sequencers. Today's sound cards, introduced in countless numbers since 1989, follow this concept.

Exploiting the ever-increasing processing power of today's computers, the next evolutionary step for the synthesizer is the software synthesizer, which runs as an application on a host computer.

The sound card (or built-in audio hardware) is needed these days only for audio input and output. The actual process of sound generation, effects processing, recording, and sequencing is performed by your computer's CPU—using the Logic Express software and instrument collection.